Session 3aABa

Animal Bioacoustics and Signal Processing in Acoustics: Bioinspiration and Biomimetics in Acoustics III

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

Chair's Introduction—8:00

Invited Papers

8:05

3aABa1. Biomimetic sonar echo parameters form cognitive maps. Roman B. Kuc (Elec. Eng., Yale, 15 Prospect St., 511 Becton, New Haven, CT 06511, roman.kuc@yale.edu)

A biomimetic audible sonar probes 2.5D targets and processes binaural echoes to extract values of eight parameters to generate two-dimensional cognitive maps. Targets are configured using posts connected by tangential planes. Being tuned to recognize posts and planes, the sonar produces a cognitive map that is composed of these two components. A platform with translational and rotational degrees of freedom employs right-ear dominance to implement a landmark-centric scanning trajectory whose step size adaptively changes with echo information. The sonar tracks the target by maintaining a constant first echo arrival time and equalizes binaural echo times to form singular echoes. When observed, singular echoes identify landmarks defined by post radii and locations. The mapping process employs five states from detection to termination that passes through the singular echo state. Separate states detect post pairs that exhibit echo interference and planes that exhibit echo amplitude differences. The scanning process terminates when the current landmark parameters match those of the first landmark. Two targets configured with three posts and an added plane illustrate the procedure.

8:25

3aABa2. The acoustic world of odontocete biosonar and technical sonar. Michael J. Roan, Rolf Müller, and Hyeon Lee (Mech. Eng., Virginia Tech, 111 Randolph Hall, 460 Old Turner St., Blacksburg, VA 24061, mroan@vt.edu)

Bats, toothed whales (odontocetes), and man-made sonar all use the same acoustical principles of echo-location. The fundamental components of these systems include transmitter frequency response, tailored waveforms, properties of a propagation medium, clutter, background noise, target strength, and receiver properties. Across all of these elements, there are very large differences (i.e., underwater versus in-air propagation). This gives rise to several interesting questions about the performance of these sonar systems. For example, given the bandwidth and duration of transmitted signals for bats and odontocetes, what are the relative target strengths/maximum imparted Doppler shifts for common prey types? How does the ambiguity function of the echo compare to these target sizes and relative Doppler shifts? What effect does the medium has and how do these systems adapt to clutter, interferers, and noise? This talk will focus on the underwater aspects of these systems which include (i) the properties of the underwater propagation medium, (ii) the geometry and material of the boundaries that limit the underwater propagation channel (this includes targets of interest and clutter), and (iii) the time-frequency and spatial properties of the underwater sources.

8:45

3aABa3. Acoustic reflectivity of a harbor porpoise *Phocoena phocoena.* Whitlow Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96744, wau@hawaii.edu), Ronald Kastelein, and Lean Helder-Hoek (SEAMARCO, Inc., Hardewick, The Netherlands)

Acoustic backscatter measurements were conducted on a stationary harbor porpoise (*Phocoena phocoena*) under controlled conditions. The measurements were made with the porpoise in the broadside aspect using three different types of signals as follows: (1) a 475-ms linear frequency-modulated (FM) pulse with a frequency range from 23 to 160 kHz; (2) a simulated bottlenose dolphin (*Tursiops truncates*) click with a peak frequency of 120 kHz; and (3) a simulated killer whale (*Orcinus orca*) click with a peak frequency of 60 kHz. The measurement with the FM pulse indicated that the mean target strength at the broadside aspect decreased from -26 dB to -50 dB as the frequency increased from 23 to 120 kHz in a nearly linear fashion (on a logarithm plot) and dropped rapidly by 27 dB to -77 dB at 150 kHz. Target strength variation with the frequency was similar to that in a previous backscatter measurement performed on a bottlenose dolphin over a similar frequency range (23–80 kHz). The target strength of the smaller harbor porpoise was about 15–16 dB lower than that of the bottlenose dolphin. The difference in the lung volume of the two species when expressed in dB was also approximately 15 dB. The results suggest that the dolphin bubbrt had broadband anechoic properties.

9:05

3aABa4. Linear and non-linear features of teleost acoustic communication and implications in software defined communication protocols. Cameron A. Matthews (Panama City Div., Naval Surface Warfare Ctr., 110 Vernon Ave., Panama City, FL 32407, cameron.matthews@navy.mil)

Many teleost species communicate via acoustic paths. Many times these paths are chosen through evolution in a way that scientists find contrary to practical physics associated with the ocean environments they are born from. This leads to a practical question of why animals would not naturally become the best basis for acoustic communication, particularly in the emerging market of software defined radios and their application in undersea acoustics. We propose a grouping of three distinct acoustic communication "styles" in terms of linearity—periodic linear, periodic non-linear, and aperiodic non-linear—and what the implications are for a dynamic and relatively wide band software defined communication system are.

9:25

3aABa5. Developing a biomimetic acoustic deterrent to reduce bat mortalities at wind turbines. Michael Smotherman (Biology, Texas A&M Univ., 3258 TAMU, College Station, TX 77843-3258, smotherman@tamu.edu), Paul Sievert, Zara Dowling (Dept. of Environ. Conservation, Univ. of Massachusetts, Amherst, MA), Dan Carlson, and Yahya Modarres-Sadeghi (Mech. Eng., Univ. of Massachusetts, Amherst, MA)

High bat mortalities at wind turbines have emerged as an unexpectedly severe environmental impact of developing wind farms all over the world, motivating an urgent need to develop effect deterrent strategies that can be implemented efficiently on a large scale. Echolocating bats use ultrasonic sonar to navigate, and some studies have shown that broadband ultrasonic noise can have a deterrent effect under certain conditions. Thus, ultrasonic noise may be an attractive approach, but attempts to incorporate electronic sound generating solutions have generally failed because of insufficient bandwidth and intensity as well as their sensitivity to the harsh environmental conditions. To get around these constraints, we designed a biomimetic whistle loosely modeled after the bat larynx that could be mounted on the moving turbine blades to passively generate ultrasonic sounds tuned to the acoustic parameters of bat's auditory system. The whistle produces multi-harmonic tones detectable by the most impacted bat species from distances approaching 100 me away. To test whether the whistle actually deters bats or alters their flight paths, we conducted a series of playback studies in the lab and field using microphone arrays and videography. This presentation will focus on the results of these behavioral studies assessing whether or not bats change their flight trajectories in response to hearing the acoustic stimulus produced by the whistle.

Contributed Paper

9:45

3aABa6. Computer simulation of energy efficient speech production during locomotion over water without regular swimming strokes. Amitava Biswas (SHS Lab., Speech and Hearing Sci., Univ. of Southern Mississippi, 118 College Dr. #5092, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Many aquatic animals need to live extended periods of time over water without much concern for fatigue. In contrast, humans are usually prone to fatigue during locomotion over water and unlikely to sustain energy efficient production of speech simultaneously. According to computer simulations, a strategy of backfloating and articulating all four or some of the limbs like oars can move the body efficiently, and the individual is less likely to fatigue and more able to maintain uninterrupted normal speech conversation for a greater period of time. The motion can be easily directed towards the feet to avoid striking the head against an obstacle when distracted by the speech conversation.

10:00–10:30 Panel Discussion

Session 3aABb

Animal Bioacoustics: Animal Bioacoustics Poster Session

Joseph Sutlive, Cochair
Translational Biology, Medicine, and Health, Virginia Polytechnic Institute and State University, 3719 Parliament Rd., Apt. 22, Roanoke, VA 24014

Ruihao Wang, Cochair
VT, 112 Hearthstone Dr, Apt. 210, Blacksburg, VA 24060

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

Contributed Papers

3aABb1. Investigating Atlantic bottlenose dolphin mismatch negativity response to pure tone stimuli. Brittany A. Romanchek and Peter M. Scheifele (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., P.O. Box 670379, Cincinnati, OH 45267, huttonba@mail.uc.edu)

Marine mammals rely heavily on acoustic cues to survive but are susceptible to hearing impairment due to anthropogenic noise. The quality of life and well-being of marine mammals under professional care is based on enrichment through training and communication conspecifics and zoo staff. Mismatch Negativity (MMN) is an electrophysiological exam that records synchronized firing of neurons from the auditory cortex and frontal lobe, allowing for assessment of echoic memory and sound differentiation capability. The objective of this study is to conduct MMN exams, with acoustic stimuli, on the Atlantic Bottlenose Dolphin (*Tursiops truncatus Montagu*) to study echoic memory and develop auditory cognition assessments for marine mammals in captivity. Three male Atlantic Bottlenose dolphins were tested in water at the Indianapolis Zoo. The oddball paradigm in this study included a frequency difference between two pure-tone stimuli (standard 2000 Hz, deviant 500 Hz). The results suggest that a similar MMN waveform morphology and latency (between 100 and 300 ms) are seen in dolphins when compared to humans. Average P1, N1, P2, and MMN values were determined for each dolphin. Developing MMN tests for animals under professional care will allow for application to marine mammals in natural habitats, which are exposed to high levels of anthropogenic noise.

3aABb2. What is a scream? Acoustic characteristics of a human call type. Jay W. Schwartz, Jonathan W. Engelberg, and Harold Gouzoules (Dept. of Psych., Emory Univ., 36 Eagle Row, Atlanta, GA 30322, jwschwa@emory.edu)

Recent research suggests that human screams comprise an innate call type, yet the defining acoustic structure of screams has not been determined. In this study, participants listened to 75 human vocal sounds, representing both a broad acoustical range and array of emotional contexts, and were asked to classify each as either a scream or a non-scream. Participants showed substantial agreement (Fleiss' kappa test, $\kappa = 0.62$) on which sounds were considered screams. To characterize the acoustic structure of screams, we measured the stimuli on 28 acoustic parameters. PCA and GLMM analyses indicated that the classification as a scream was positively correlated with 3 acoustic factors: one corresponding to high pitch and roughness, another corresponding to a wide fundamental frequency range and narrow bandwidth, and a third negatively correlated with roughness. Thus, the role of roughness in identifying screams appears to depend in part on pitch. Twenty-six stimuli were agreed upon by $> 90\%$ of participants to be screams, but others evoked mixed responses, suggesting that while screams might represent an innate call type, their acoustic structure is not narrowly fixed, perhaps reflecting the range of emotions and contexts in which humans employ them. The evolutionary significance of scream acoustics is discussed.

3aABb3. Discrimination of ripple spectra in a bottlenose dolphin in quiet and after noise exposure. Dmitry Nechaev, Vladimir Popov (Inst. of Ecology and Evolution, Moscow, Russian Federation), Alexander Supin, Mikhail Tarakanov, and Evgeniya Sysueva (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, evgeniasysueva@gmail.com)

The frequency resolving power (FRP) of hearing in quiet and after noise exposure was measured in a bottlenose dolphin using rippled-spectrum test stimuli and noninvasive recording of rhythmic evoked responses (the rate following response, RFR) to ripple phase reversals. Both the test signal and noise had band-limited spectra with the same central frequency; however, the noise had a non-rippled spectrum. The noise level was from -20 to 10 dB re test signal level. The baseline ripple density resolution depended on signal level and was the highest at levels from 80 to 100 dB re 1 μPa. At signal levels both above (up to 130 dB re 1 μPa) and below (down to 80 dB re 1 μPa) the optimal level, the ripple density resolution decreased. The impact of noise was different for different test signal levels. For low test signal levels (70 to 100 dB re 1 μPa), noise decreased RFR magnitude and resolution, whereas for high test signal levels (110 to 130 dB re 1μPa), low-level noise increased RFR magnitude and resolution. [Work supported by the Russian Science Foundation (Project 17-74-20107) awarded to E.V.S.]

3aABb4. Penguin vocalization classifications. Alexandra M. Parshall and Peter M. Scheifele (Univ. of Cincinnati, #31, 2147 Madison Rd., Cincinnati, OH 45208, parshaam@mail.uc.edu)

The present study researches five different penguin species vocalizations who currently residing at the Newport Aquarium in Newport, KY. These penguins—while different species—coexist in the same habitat at the Newport Aquarium. It was purposed that the different species do not "speak the same language" and therefore do not respond to different species of penguin's calls, despite sharing the same enclosure. Therefore, this study aimed at collecting and categorizing the different species' calls and then proceeding to analyze the vocal data to see if any of the species shared similar vocal spectrum similarities. Once collected, the aim of this data is twofold: primarily to classify the different penguin species calls—especially during specific events (i.e., defending a nest, distress, calling for a mate, etc.) and secondarily to analyze the vocal spectral energy of the different species to see if similarities exist.

3aABb5. Analysis of BAER tests of *Tursiops truncatus Montagu* in comparison of captive males to naval trained males in the open ocean. Elizabeth Hanson and Peter M. Scheifele (Univ. of Cincinnati, 3230 Eden Ave., Cincinnati, OH 45342, hansonea@mail.uc.edu)

Do captive Atlantic Bottlenose Dolphins, *Tursiops truncatus Montagu*, have impaired hearing? This study compared the AEP results from captive dolphins to those who are trained by the Navy in the open ocean. The dolphins removed from the wild and relocated to captive environments could have a reduced hearing range as a result of the transfer to an area with potentially increased baseline noise levels. Auditory Brainstem Response (ABR) tests were conducted with underwater acoustic stimuli projection and reception in both the captive and the Navy dolphins. *Tursiops truncatus Montagu* has a range of hearing from 0.1-120 kHz. The hearing of dolphins that were born in captivity was closest to the naval dolphins in the ocean compared to those who were rescued from the wild. Captive dolphins were exposed to a constant 0.1-kHz noise levels for a brief period of time. This may potentially have had an effect on the latency of the peaks I-V and the correlating troughs in their ABR waveform. The results of this study can be used to assess auditory health of marine mammals and to determine the effect of anthropogenic noise levels whether in captive or natural marine mammal environments.

3aABb6. Analysis of bioacoustics and noise pollution in the penguin exhibit at Newport Aquarium. Julie Colaianni and Peter M. Scheifele (Audiol., Univ. of Cincinnati, 3216 Glendora Ave., Cincinnati, OH 45220-2206, colaiaja@mail.uc.edu)

The present study researches the acoustic habitat of the penguin exhibited at Newport Aquarium in Newport, KY by means of measuring environmental noise within their exhibit. Noise exposure has been well studied in humans by using a weighted decibel scale to provide safety guidelines for the level of noise and the maximum time one can be exposed to each level. Therefore, the same principle is being used to observe if these animals are experiencing noise pollution from their own environmental sounds. The absorption coefficients of the materials that make up the habitat were also taken into consideration during data collection. The aim of this research is to gather calculations for the time it takes for the sound pressure level to reduce by 60dB (RT60) after periods of vocalization by the colony of penguins and to determine if these reverberation values reach the peak sensitivity of hearing thresholds for them.

WEDNESDAY MORNING, 15 MAY 2019 NUNN, 9:00 A.M. TO 11:55 A.M.

Session 3aBA

Biomedical Acoustics, Signal Processing in Acoustics, and Physical Acoustics: Interaction of Light and Ultrasound I

E. Carr Everbach, Cochair
Engineering, Swarthmore College, 500 College Avenue, Swarthmore, PA 19081

Ronald A. Roy, Cochair
Dept. of Engineering Science, University of Oxford, 30 Edgeway Road OX3 0HD, United Kingdom

Invited Papers

9:00

3aBA1. Photoacoustic imaging in cancer medicine and research: Systems, results and future directions. Jeffrey C. Bamber (Joint Dept. of Phys., Inst. of Cancer Res. and Royal Marsden Hospital, 6 Sandbourne Ave., Merton Park, London SW19 3EN, United Kingdom, jeffrey.bamber@physics.org)

This paper reviews selected previous work in photoacoustic imaging conducted at the Institute of Cancer Research or with collaborators and considers directions for future research. For flexible introduction as an additional mode to clinical freehand ultrasound scanning, a multispectral epiphotoacoustic system was built by adapting a Zonare z.oneUltra™. Early clinical findings from photoacoustic imaging of breast tumours were consistent with contrast MRI although also provided evidence that photoacoustic image interpretation and penetration were limited by clutter generated by photoacoustic emissions from regions at or near the skin surface. Schemes for improving photoacoustic signal-to-clutter ratio were therefore explored. These have included deformation compensated averaging, spatial coherence processing, photoacoustic contrast agents, and localised vibration tagging (LOVIT). LOVIT (implemented using acoustic radiation force) was particularly successful, tripling penetration depth in breast-mimicking phantoms. It has also been implemented on a Verasonics™ platform and extended using a 3 + 3 interleaved comb-push method, which further reduces the clutter and improves the frame rate. Multidimensional and other extensions to LOVIT remain possible. Meanwhile, work at other institutions has explored various propagation model-based schemes. Most of the clutter reduction methods are complementary in nature, and a combined approach therefore represents a worthwhile direction for the future work.

9:20

3aBA2. Synergy of light and sound for deep-tissue optical imaging and focusing. Puxiang Lai (Biomedical Eng., Hong Kong Polytechnic Univ., ST412, BME, Hung Hom, Kowloon, Hong Kong, puxiang.lai@polyu.edu.hk)

Light and sound are the two most dominant mechanisms with which we naturally perceive this world. Both have their significant impacts yet with inherent limitations. For example, sound is insensitive to soft tissue functional changes, and light is strongly scattered in tissue, resulting in a trade-off between penetration depth and resolution. Here, we present our most recent research efforts of exploiting the synergy of light and sound to overcome this limitation. Specifically, photoacoustics converts diffusive photon into non-scattering ultrasonic waves, enabling a high-contrast sensing of optical absorption with ultrasonic resolution in deep tissue, overcoming the optical diffusion limit from the signal detection perspective. The generation of photoacoustic signals, however, is still throttled by the attenuation of photon flux due to the strong diffusion effect of light in tissue. Therefore, wavefront shaping is introduced, so that multiply scattered light could be manipulated so as to retain optical focusing or sufficient photon flux even at depths in tissue. We will present the recent development of photoacoustic imaging and optical wavefront shaping in our lab. Potential applications, existing challenges, and further improvement are also discussed.

9:40

3aBA3. Irreversible shifts in optical autofluorescence spectra applied to the assessment of thermal lesion formation under high intensity focused ultrasound. Shamit Shrivastava (Dept. of Eng. Sci., Oxford Univ., Oxford, Oxfordshire OX1 3PJ, United Kingdom, shrivastava.shamit@gmail.com), E. Carr Everbach (Eng., Swarthmore College, Swarthmore, PA), Jason L. Raymond, and Ronald Roy (Eng. Sci., Oxford Univ., Oxford, United Kingdom)

High-intensity focused ultrasound (HIFU) is often used to create lesions, or regions of tissue destruction due to heating and cavitation activity, most often in tumors or other diseased tissues. However, the acoustic properties of tissues denatured by heat are not very different from those of untreated tissue, making lesion detection and quantification difficult by ultrasound alone. Autofluorescence refers to the broadband emission of light within materials that are stimulated by narrowband incident light, typically from a laser. It is a common characteristic of lipids, proteins, and other biomolecules, and the "autofluorescence spectrum" is a function of the state of the material. We have examined irreversible shifts in the dominant autofluorescence spectra of proteins denatured by HIFU heating. These shifts appear to be related to protein conformational changes due to denaturation. We report on the feasibility of using optical autofluorescence as a means of quantifying *in vitro* lesion formation by HIFU for optically accessible tissues.

10:00–10:15 Break

10:15

3aBA4. Spectroscopic photoacoustic imaging for cardiovascular interventions. Sophinese Iskander-Rizk (Biomedical Eng., Erasmus MC, WYTEMAWEG 80, Rotterdam 3015CN, The Netherlands, s.iskander-rizk@erasmusmc.nl), Pieter Kruizinga (Neurosci., Erasmus MC, Rotterdam, The Netherlands), Min Wu (Biomedical Eng., TU Eindhoven, Rotterdam, The Netherlands), Antonius F. W. van der Steen, and Gijs Van Soest (Biomedical Eng., Erasmus MC, Rotterdam, The Netherlands)

Alongside ultrasound imaging, photoacoustic imaging (PA) augments the imaging system with tissue composition information. Indeed, PA primarily maps tissue optical absorption and thus can reveal tissue composition. Localizing and identifying specific biomolecules can be very useful for both diagnostic imaging and evaluation of the treatment effect in various applications. PA imaging can for instance discern lipid-rich atherosclerotic plaques from fibrotic ones. It can also distinguish highly vascularized or highly oxygenated tissue from hypoxic areas in tumors. However, multiple parameters affect the PA signal received, rendering a direct mapping from signal received to absorption challenging. For instance, at selected wavelengths, multiple chromophores may contribute to the signal observed. In order to separate the different chromophores, judicious spectral tuning of imaging wavelengths can help. We call this technique spectroscopic photoacoustic imaging (sPA). Based on sPA imaging, we were capable to super-localize sources down to 1/20th of the imaging system point spread function (PSF). sPA can also improve the imaging specificity and sensitivity of targeted tissue features. In this talk, we showcase the many benefits of sPA imaging for image enhancement. We also discuss challenges and feasibility to integrate in surgical tools for minimally invasive interventions. In particular, we focus on how sPA studies can take the specific problem of visualization of RF ablation for atrial fibrillation from bench to bedside.

10:35

3aBA5. Towards photoacoustic sensing and mapping of neuroelectrical and hemodynamic activity. Parag V. Chitnis (Dept. of Bio-Eng., George Mason Univ., 4400 University Dr., 1G5, Fairfax, VA 22032, pchitnis@gmu.edu)

Understanding the neurological function and disorders is an enduring challenge, particularly because the brain activity involves a diverse set of chemical, ionic, and electrical interactions spanning a wide range of spatial scales from microns to several centimeters. Recording these time-varying dynamics in intact, mammalian brain provides an insight into how signaling is processed in neural networks and how these signals modulate physiological function. Optical-fluorescence techniques have emerged as tools of choice for the imaging neuronal activity. Despite significant advances in fluorescent voltage and calcium reporters, these methods are limited to penetration depths of less than 1 mm. One viable alternative to overcome the depth limitation is photoacoustic sensing, which relies on absorption of light and subsequent thermoelastic generation of ultrasound. Photoacoustic methods provide spectroscopic specificity to endogenous and exogenous chromophores, but the molecular information is relayed to the sensor acoustically, which is not as susceptible to scattering in tissue as light. We will present on ongoing efforts to develop (1) PA-based voltage reporters, (2) photoacoustic imaging of the voltage and hemodynamic activity at micro- and meso-scales, and (3) a deep learning framework for achieving super-resolution (exceeding the diffraction limit of ultrasound) photoacoustic tomography of the brain. [Partially supported by NIH-1R21EY023012.]

10:55

3aBA6. Super-resolution approaches in photoacoustic imaging. Bastien Arnal, Sergey Vilov, Guillaume Godefroy (LiPhy, Université Grenoble Alpes, CNRS, Grenoble F-38000, France, bastien.arnal@univ-grenoble-alpes.fr), and Emmanuel Bossy (LiPhy, Université Grenoble Alpes, CNRS, Saint-Martin d'Hères, France)

The resolution of photoacoustic imaging (PAI) is limited at depths by the diffraction limit. Several ways have been introduced to achieve super-resolution. In the context of imaging the vasculature, the presence of flow can be exploited in two regimes, distinct by the concentration of flowing absorbing particles. In the high concentration regime, we proposed to exploit the absorption fluctuation caused by flowing absorbers by analyzing nth-order statistics of temporal signal fluctuations. In the low concentration regime, when absorbers appear one-by-one in each acoustic resolution spots, the localization microscopy technique can be adapted to our problem. While these two methods improve the resolution greatly, their cost is to reduce temporal resolution, because of the need to record thousands of images. Supposing the knowledge of the PSF (point spread function) of the imaging system, it is possible to the regain temporal resolution. After the simulation of the forward model, the imaged object can be recovered by solving a minimization problem. We will show that adding a sparsity constraint to this problem can enhance the resolution. These techniques have been investigated in both simulations and experiments in microfluidic channels. Such super-resolution approaches bring the optical contrast at depth closer to the cellular level.

11:15

3aBA7. Sono-photoacoustic theranostics using phase-changing contrast agents. David S. Li, Kacper Lachowski (Dept. of Chemical Eng., Univ. of Washington, 105 Benson Hall, Box 351750, Seattle, WA 98195, dsli@uw.edu), Ivan Pelivanov (BioEng., Univ. of Washington, Seattle, WA), Thomas Matula (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), Matthew O'Donnell (BioEng., Univ. of Washington, Seattle, WA), and Lilo Pozzo (Chemical Eng., Univ. of Washington, Seattle, WA)

Phase-change contrast agents are low boiling point liquid perfluorocarbon droplets that can be vaporized to form larger microbubbles. Droplet vaporization can be used for both imaging and therapy. Sono-photoacoutics is a non-linear imaging method that uses simultaneous optical and acoustic pulses to activate phase-change contrast agents. By combining photothermal heating from a laser pulse and the negative pressure from an acoustic pulse, lower droplet activation thresholds are achieved than from either source alone. In this study, we demonstrate in an *in vitro* model that sono-photoacoutics activation of polypyrrole coated perfluorocarbon droplets can be used to disrupt 2-cm long fibrin clots to restore flow. Polypyrrole coated droplets under 200 nm in diameter were introduced upstream and allowed to diffuse into the clot. A 1.24-MHz single element transducer coaxially aligned with a 1064-nm pulsed laser was used to scan the clot, activating any agents within the clot. Agent activation was quantified using passive cavitation detection, while flow was monitored using a digital balance. Our results show that the droplets can freely diffuse into fibrin clots and SPA activation can restore approximately 25% of the flow.

11:35

3aBA8. Light, sound, nanobubbles: New approach to contrast-enhanced ultrasound and photoacoustic imaging. Stanislav Emelianov (ECE and BME, Georgia Inst. of Technol., 777 Atlantic Dr., Atlanta, GA 30332-0250, stas@gatech.edu)

To overcome the most significant deficiencies of conventional and contrast-enhanced ultrasound imaging—low contrast and large size of microbubbles, we introduced new class of contrast agents—nanometer scale particles that are capable of escaping vascular compartments, penetrating into tissue, and then, once they reach the target site, generating sufficient ultrasound and photoacoustic contrast upon user-controlled optical activation. These multimodal contrast agents—phase-change perfluorocarbon nanodroplets and plasmonic nanoparticles covered by azide compounds—are stable at physiological temperatures, biocompatible, and monodisperse in size. Given the unique properties of the particles, our approach to image these particles is drastically different and is based on ultrasound read-out of the optically induced temporal changes. Specifically, time-varying ultrasound signals exhibited by the nanoparticles versus the static background are used to reconstruct a high contrast, background-free image of the contrast agent. Furthermore, these particles allow for multiplexed molecular imaging by permitting user-controlled triggering of distinct color-coded populations of contrast agents via tuning of the incident laser irradiation to match peak optical absorption of the particles. Finally, nanoparticles may also contain therapeutic cargo and thus can be used for controlled drug delivery and release. This presentation, via examples, will discuss diagnostic imaging and image-guided therapy using the gas-generating nanoparticles.

Session 3aCA

Computational Acoustics, Biomedical Acoustics, Physical Acoustics, Underwater Acoustics, Signal Processing in Acoustics and Structural Acoustics and Vibration: Finite Difference Time Domain Method Across Acoustics

Michelle E. Swearingen, Cochair
U.S. Army ERDC, Construction Engineering Research Laborartory, P.O. Box 9005, Champaign, IL 61826

Jennifer Cooper, Cochair
Johns Hopkins University Applied Physics Laboratory, 11100 Johns Hopkins Rd, Mailstop 8-220, Laurel, MD 20723

Subha Maruvada, Cochair
U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993

Invited Paper

9:00

3aCA1. Applications of finite-difference time-domain for architectural acoustics consulting. Laura C. Brill and John T. Strong (Threshold Acoust., 141 W. Jackson Blvd., Ste. 2080, Chicago, IL 60604, jstrong@thresholdacoustics.com)

Acoustics consultants have many tools in their arsenal to evaluate and design rooms and architectural elements; the computational resources available to this point have made the use of wave-propagation models impractical for the common user. Threshold acoustics has found it both useful and now computationally feasible to supplement more traditional, geometric analysis with the simulation of wave-propagation using finite-difference time-domain (FDTD). Our group has developed first-order leapfrog FDTD routines in MATLAB for simulating wave propagation in an isotropic medium in two- and three-dimenision with perfectly matched layers being the boundary condition. The placement of solid elements within the test space allows analysis of arbitrary geometries. For additional computational power, our group has utilized GPU computing clusters available through Amazon Web Services accessed directly through MATLAB. Our method is based on the simulation of an impulse response and subsequent analysis of the impulse response consistent with traditional *in situ* testing methods. Applications to date include analysis of the scattering behavior of acoustically shaped surfaces and evaluation of the array behavior of architectural reflector panels.

Contributed Papers

9:20

3aCA2. Small room auralizations: Investigating hybrid methods of acoustic simulations utilizing wave field synthesis. E. K. Ellington Scott and Jonas Braasch (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, scotte3@rpi.edu)

Auralizations have become a beneficial tool in acoustic design of performance spaces, but nearly all research has been limited to Western classical music venues. Other venues, for example, jazz clubs, pose very different auralization challenges because their dimensions are typically smaller, and pure geometrical methods can no longer be applied. With a dramatic increase in computational performance, an effort has been placed in integrating geometric and wave-based models in auralization simulations that are suitable to auralize smaller venues. Finite difference time domain methods are applied for low-frequency numerical analysis of the Dizzy's Club Coca-Cola and The Village Vanguard. This research endeavors to establish a method in identifying the crossover frequency between the geometric and wave-based models, as it pertains to small room acoustics to recreate the sound of jazz venues using the wave field synthesis. The 128-channel wave field synthesis system of Rensselaer's Collaborative Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab) is used to perceptually evaluate the hybridized auralizations. Room acoustical measurements are obtained within the rendered sound field and compared to the measurements of the original locations for an objective analysis.

9:35

3aCA3. Finite difference time domain ray-based modelling of acoustic scattering for target identification and tracking. Grant Eastland (Test & Evaluation, Naval Undersea Warfare Ctr. Div. Keyport, 610 Dowell St., Keyport, WA 98345, grant.eastland@navy.mil)

The Finite Difference Time Domain (FDTD) method has provided a powerful technique for modelling and simulation of solutions of a variety of acoustics problems. The purpose of this investigation is to present work on the development of time-domain models of acoustic scattering from targets near a flat pressure-release boundary for use in identification and tracking from a moving receiving platform. The timing from the acoustic source to reception is dependent on the location dependent sound speed profile and the specular scattering points on the target. There can be multiple specular points revealed on the target because of interactions with the flat boundary, each can be modelled with the FDTD method and providing the correct path corrections using the same method to account for the variation in propagation path brought on by the sound speed profile of the propagating environment.

9:50

3aCA4. Acoustic wave scattering from dynamic rough sea-surfaces using the finite-difference time-domain method. Alex Higgins and Martin Siderius (Elec. & Comput. Eng., Portland State Univ., 1900 SW 14th Ave., Ste. 25-01, Portland, OR 97201, higginja@ece.pdx.edu)

Numeric models for underwater acoustic propagation typically assume the sea-surface to be either perfectly smooth or rough but "frozen" in time. For long sonar signals on the order of tens of seconds, the sea-surface can interact at many different wave displacements over its duration. This causes anomalies in the received signal which introduces additional transmission losses and Doppler effects. The impact of including roughness and motion of the sea-surface on sonar systems is investigated using the finite-difference time-domain (FDTD) method. The FDTD method is a numeric technique that is well suited for modeling boundary roughness and motion. This is due to its ability to directly configure complex boundary conditions in the surrounding simulation grid and full pressure wave propagation in the time-domain. The rough, time-evolving sea-surface is modeled using a Pierson-Moskowitz (PM) frequency spectrum which is simple to implement and defined using just wind speed and direction. The results from FDTD simulations of static rough sea-surfaces are compared to a previously established integral solution method to evaluate the validity of the approach. Agreement is also demonstrated for FDTD simulations of a dynamic rough sea-surface and a theoretic statistical model. [Work supported by the Office of Naval Research.]

10:05–10:20 Break

10:20

3aCA5. Time-domain simulations of sound propagation near a ground surface in comparison with the ANSI impedance measurement models. Z. C. Zheng, Junjian Zhang (Aerosp. Eng., Univ. of Kansas, 2118C Learned Hall, 1530 W 15th St., Lawrence, KS 66045, zzheng@ku.edu), and Guoyi Ke (Mathematics and Physical Sci., Louisiana State Univ. of Alexandria, Alexandria, LA)

Time-domain simulations with the Zwikker-Keston porous-material model are carried out on the geometries specified in the ANSI models for determining the acoustic impedance of ground surfaces (ANSI/ASA S1.18). The comparisons between the simulation results and the results provided in ANSI show very good agreement when the flow resistivity of the ground material is high. The effect of ground roughness is investigated by adding periodic spacing triangular strips on the ground. The numerical method for simulating rough ground surfaces, which combines the time-domain simulation with an immersed-boundary method, is validated by comparing with experimental data in the literature. It is found that when the ground roughness is introduced to the ANSI geometries, the predicted sound level difference between the two microphones in the ANSI geometries tends to shift towards lower frequency ranges with the rough ground, even though the roughness is within the allowed roughness height specified in the ANSI models.

10:35

3aCA6. Application of Elastodynamic Finite Integration Technique (EFIT) to three-dimensional wave propagation and scattering in arbitrary geometries. Sean Raley and Eric A. Dieckman (Mech. Eng., Univ. of New Haven, 300 Boston Post Rd., West Haven, CT 06516, srale1@unh.newhaven.edu)

Over several decades, railroad ultrasonic examination (UE) industry techniques have primarily been developed through simple analytical modeling and experimental approaches. However, with present-day computational capabilities, we can use numerical techniques such as the Elastodynamic Finite Integration Technique (EFIT) to fine-tune systems for complex applications before the fabrication process begins. EFIT is well-established as a useful method in numerical analysis of ultrasonic wave propagation with distinct advantages over the Finite Difference Time Domain method. Several software packages exist that use EFIT as the primary method for simulating the behavior of ultrasonic waves over time in two or three dimensions, but none of them are well-suited for railroad UE R&D. This paper explores the development of a tool developed for this purpose which was designed to: (1) allow for the input of various profile geometries, boundary conditions, and material inclusion geometries (such as a bolt hole in a railroad track); (2) allow for the input of specific ultrasonic impulses from varying emitter designs; and (3) produce verifiable results, as confirmed by experimental measurements.

Invited Paper

10:50

3aCA7. Estimation of parameters quantifying porosity in random porous structures using ultrasonic attenuation: Solving the inverse problem. Omid Yousefian (North Carolina State Univ., Raleigh, NC), Rebekah White (North Carolina State Univ., Chapel Hill, NC), H. T. Banks, and Marie M. Muller (North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

The goal of this study is to estimate the porosity (pore size and density) of numerically simulated random porous three-dimensional structures mimicking simplified geometries of cortical bone using attenuation of ultrasonic waves in the MHz range. To do so, we use a physics-based model derived from the Independent Scattering Approximation (ISA) to mathematically model the attenuation of elastic waves in porous structures as a function of the parameters that we wish to identify: pore size and density. Frequency-dependent attenuation data are generated using three-dimensional finite-difference time-domain simulations in the range of 1–8 MHz in mono-disperse structures with the pore diameter and density ranging from 100 to 200 μm and 20–50 pore/mm^3, respectively. We then solve an ordinary least squares (OLS) inverse problem to recover the pore size and density by minimizing the sum of squared errors between the simulated data and the model prediction. In doing so, we demonstrate that we can estimate with confidence the parameters quantifying porosity using the ISA model given three-dimensional numerically simulated attenuation data.

Session 3aED

Education in Acoustics: Hands-On Demonstrations

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

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomena. In this session, "Hands-On" demonstrations will be set-up for a group of middle- and high-school students from the Louisville area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students' educational development and is part of the larger "Listen Up" education outreach effort by the ASA. Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Any acousticians wanting to participate in this fun event should e-mail Keeta Jones (kjones@acousticalsociety.org).

Session 3aMU

Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Polyphonic Pitch Perception and Analysis I

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

M. Torben Pastore, Cochair
Architectural Acoustics, Rensselaer Polytechnic Institute, 4 Irving Place, Troy, NY 12180

Chair's Introduction—8:00

Invited Papers

8:05

3aMU1. Pitch perception of concurrent high-frequency complex tones. Daniel Guest and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 E River Rd., Minneapolis, MN 55455, guest121@umn.edu)

Accurate pitch perception is possible for harmonic complex tones with fundamental frequencies (F0s) in the musical range (e.g., 1.4 kHz) but with all harmonics beyond the putative limits of phase locking. However, it is unknown whether pitch perception in more complex scenarios, such as with concurrent complex tones, is possible using these stimuli. To address this question, we measured (1) F0 difference limens (F0DLs) and (2) target-to-masker ratios (TMRs) required to detect a fixed F0 difference in a mixture of complex tones with low F0s (∼280 Hz) or high F0s (∼1400 Hz). The target tones were filtered to ensure that in the high-F0 case, only harmonics beyond the limits of phase locking were present. Pitch perception was poorer for isolated high-F0 tones than for isolated low-F0 tones and adding a masker complex tone with a geometrically centered F0 impaired performance for both high-F0 and low-F0 tones. The TMRs required to achieve good performance in the presence of two complex tone maskers were higher for high-F0 tones than for low-F0 tones. The results should help determine whether different mechanisms underlie the perception of combinations of complex tones at low and high frequencies. [Work supported by Grants NIH R01 DC005216 and NSF NRT-UtB1734815.]

8:25

3aMU2. Multi-pitch spike representation using a Finite-Difference Time Domain (FDTD) cochlear model. Rolf Bader (Inst. of Systematic Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

The spike representation of multi-pitch sounds leaving the cochlear is discussed using a Finite-Difference Time Domain (FDTD) physical model of the cochlear with musical sounds as input. Previously, the model has lead to a new pitch theory, where it was found that the interspike intervals (ISI) of the fundamental periodicity of a single-pitch sound are present at multiple Bark bands in the cochlear. This is due to drop-outs of spikes in Bark bands of higher harmonics. Pitch is therefore represented as the most prominent periodicity within the auditory nerve. In the multi-pitch case, the model shows several cases of such multi-Bark periodicities. With multi-pitch sounds having a residual pitch, the strongest of these multi-Bark cases is that of the residual periodicity. With several pitches perceptually clearly distinguishable one from another, each pitch periodicity appears as a multi-Bark periodicity. Still with sounds where the single pitches are not so clear and separating them needs musical training and intense and multiple listening of the sound, the multi-Bark cases are blurred. Therefore, the model represents the perception of multi-pitch sounds during immediate perception and identification of multiple pitches needing musical training is no longer present at this low-level sound representation at the cochlear output, as expected.

8:45

3aMU3. Spectro-temporal templates unify the pitch of resolved and unresolved harmonics. Shihab Shamma and Kelsey J. Dutta (Univ. of Maryland, 2202 A.V. Williams, College Park, MD, sas@isr.umd.edu)

Pitch is a fundamental attribute in auditory perception that is involved in source identification and segregation, music, and speech understanding. When harmonics are well-resolved, the induced pitch is usually salient and precise; however, when the harmonics are not completely resolved, the pitch percept becomes less salient and poorly discriminated. Previous models relying on harmonic spectral templates have been able to account fully for the pitch of the resolved but not of the unresolved harmonics. I will describe a biologically motivated model of templates that combine both spectral and temporal cues to estimate both pitch percepts. Specifically, the pitch of unresolved harmonics is estimated through bandpass filters implemented by resonances in the dendritic trees of neurons in the early auditory pathway. It is demonstrated that organizing and exploiting such dendritic tuning can arise spontaneously even in response to white noise. We show how these temporal cues become integrated with those of spectrally resolved harmonics, effectively creating spectro-temporal harmonic templates for all pitch percepts. We finally discuss how this approach can account for all major monaural pitch percepts, as well as pitch percepts evoked by dichotic binaural stimuli.

9:05

3aMU4. Autocorrelation models of pitch perception: In memory of ray meddis. William Yost (Spatial Hearing Lab, ASU, P.O. Box 870102, Tempe, AZ 85287, william.yost@asu.edu) and Roy D. Patterson (Physiol., Univ. of Cambridge, Cambridge, United Kingdom)

About 60 years ago, Licklider (1951, *Experientia*) described a qualitative model of autocorrelation that appeared to explain complex pitch perception. However, it was almost 40 years later before, Slaney and Lyon (1990, *ICAASP*) and Meddis and Hewitt (1991, *JASA*) incorporated autocorrelation into computational auditory models to test Licklider's model of pitch perception. The Meddis and Hewitt frontend combined a level-dependent gammatone filterbank with Meddis' (1986, *JASA*) inner hair-cell model. The "autocorrelograms" it produced revealed the temporal regularity of simulated auditory-nerve interspike intervals that occur in response to complex stimuli. The model was/is remarkably successful in accounting for a large set of complex pitch perception data. The model emphasizes the temporal structure of the neural information produced by a complex sound, in contrast to models that use a simple spectral representation of the resolved harmonics of a complex sound. Today, those who model complex pitch perception either use an autocorrelation-like approach or argue why such an approach does not work. The Meddis-Hewitt model is almost always considered in these modeling efforts. This presentation will describe the Meddis-Hewitt model and discuss its lasting effect on the study of complex pitch perception. [Work supported by NIDCD and Facebook Reality Labs grants to WAY.]

9:25

3aMU5. Fundamental frequency (F0) discrimination of one complex tone in the presence of another: The role of excitation pattern and temporal fine structure cues. Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

The perceptual segregation of simultaneous harmonic complex tones depends partly on differences in the fundamental frequency (F0) between those tones. Human listeners have some ability to group together harmonics of a given F0 and segregate them from harmonics of a different F0. This paper reviews studies of the ability to detect changes in F0 of one complex tone in the presence of another complex tone with the same or a different mean F0. The studies include conditions where the harmonics in the complex tones would have been resolved, partially resolved, or completely unresolved. Modeling using excitation patterns and "summary autocorrelation" and "stabilized auditory image" models suggests that while excitation-pattern cues are useful for complex tones with resolved harmonics, the use of temporal fine structure (phase locking) information is required to account for the small F0DLs obtained when harmonics are barely, if at all, resolved.

9:45

3aMU6. Spectral and temporal models of human pitch perception with mixtures of three concurrent harmonic complexes. Jackson Graves (Département d'Études Cognitives, École Normale Supérieure, 29, rue d'Ulm, Paris 75005, France, grave276@umn.edu) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

In music and other everyday situations, humans are often presented with three or more simultaneous pitches, each carried by a harmonic complex tone, but few empirical or theoretical studies have addressed how pitch is perceived in this context. In three behavioral experiments, mixtures of three concurrent complexes were filtered into a single bandpass spectral region, and the relationship between the fundamental frequencies and spectral region was varied in order to manipulate the extent to which harmonics were resolved either before or after mixing. Listeners were asked to discriminate major from minor chords (Experiment 1) or to compare the pitch of a probe tone to that of a target embedded in the mixture (Experiments 2 and 3). In all three experiments, listeners performed above chance even under conditions where traditional rate-place models would not predict individually resolved components. Human behavioral results were compared to predictions from two classes of pitch model: a rate-place model using harmonic template matching and a temporal model using summary autocorrelation. Predictions from a combined model, using both rate-place and temporal information, were more accurate than predictions from either model alone, suggesting that humans may integrate these two kinds of information. [Work supported by NIH grant R01DC005216.]

10:05

3aMU7. Perceptual fusion of musical notes suggests universal representations of dissonance despite culture-dependent aesthetic associations. Malinda J. McPherson (Div. of Medical Sci., Harvard Univ., MIT Bldg. 46-4078, 43 Vassar St., 46-4078, Cambridge, MA 02139, malindamcpherson@g.harvard.edu), Sophia E. Dolan (Wellesley College, Wellesley, MA), Tomas Ossandon, Joaquin Valdes (Dept. of Psychiatry, Pontificia Universidad Católica de Chile, Santiago, Chile), Eduardo A. Undurraga (Escuela de Gobierno, Pontificia Universidad Católica de Chile, Santiago, Chile), Nori Jacoby (The Ctr. for Sci. and Society, Columbia Univ., New York, NY), Ricardo Godoy (Heller School for Social Policy and Management, Brandeis Univ., Waltham, MA), and Josh McDermott (Dept. of Brain and Cognit. Sci., Massachusetts Inst. of Technol., Cambridge, MA)

Music varies enormously across cultures, but some traits are widespread. Cross-cultural consistency in music could be driven by universal perceptual mechanisms adapted to natural sounds, but supporting evidence has been circumstantial due to the dearth of cross-cultural research. Here, we explore whether such perceptual mechanisms impose universal similarity relations on musical structure, potentially dissociating from culture-specific aesthetic judgments about music. We measured one possible signature of these similarity relations—the extent to which concurrent notes are perceived as a single sound—in members of a small-scale Amazonian society and Western listeners. We also measured aesthetic responses to the same stimuli. Unlike Westerners, Amazonian listeners were aesthetically indifferent to whether note combinations were canonically consonant (with aggregate frequency spectra resembling the harmonic series). However, Amazonians were nonetheless more likely to hear consonant combinations as a single sound, with fusion judgments that qualitatively resembled those of Western listeners. Thus, even in a culture with little exposure to Western harmony, reliance on harmonic frequency relations for sound segregation evidently induces consistent perceptual structure in note combinations. The results suggest that perceptual mechanisms for representing music can be shared across cultures, even though the perceptual equivalences that result give rise to culture-specific aesthetic associations.

10:25–10:40 Break

Contributed Paper

10:40

3aMU8. Polyphonic pitch perception in rooms using deep learning networks with data rendered in auditory virtual environments. Nathan Keil, David A. Dahlbom (Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, keiln@rpi.edu), Jeremy Stewart, Matthew Goodheart, Curtis Bahn, Mary Simoni (Arts, Rensselaer Polytechnic Inst., Troy, NY), Michael Perrone (Thomas J. Watson Res. Ctr., IBM, Yorktown Heights, NY), and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

This paper proposes methods for generation and implementation of uniform, large-scale data from auralized MIDI music files for use with deep learning networks for polyphonic pitch perception and impulse response recognition. This includes synthesis and sound source separation of large batches of multitrack MIDI files in non-real time, convolution with artificial binaural room impulse responses, and techniques for neural network training. Using ChucK, individual tracks for each MIDI file, containing the ground truth for pitch and other parameters, are processed concurrently with variable Synthesis ToolKit (STK) instruments, and the audio output is written to separate wave files in order to create multiple incoherent sound sources. Then, each track is convolved with a measured or synthetic impulse response that corresponds to the virtual position of the instrument in the room before all tracks are digitally summed. The database now contains the symbolic description in the form of MIDI commands and the auralized music performances. A polyphonic pitch model based on an array of autocorrelation functions for individual frequency bands is used to train a neural network and analyze the data [Work supported by IBM AIRC grant and NSF BCS-1539276.]

10:55

3aMU9. Matching pitch gestures to visual trajectories as a function of acoustic scales. Sven-Amin Lembke (Music, Technol. and Innovation - Inst. for Sonic Creativity (MTI2), De Montfort Univ., Clephan Bldg., Rm. 00.07b, Leicester LE1 9BH, United Kingdom, sven-amin.lembke@dmu.ac.uk)

Auditory gestures are commonly used in music and sonification, although little is known about their perceived shape. A listening experiment aimed to study how pitch-based gestures that rely on different acoustic scales are perceived and matched to analogous visual trajectories. Four scales were studied, varying in curvature on a linear time-vs-frequency plane: linear frequency in Hz, equivalent-rectangular-bandwidth (ERB) rate, an exponential function twice the curvature of ERB rate, and a logarithmic function with inverse curvature to ERB rate. The pitch-glide stimuli were based on filtered noise and spanned two octaves. These glides were tested below and above a 1-kHz anchor, ascending and descending in pitch, and for two durations, all serving as independent variables (IVs). An interactive visual trajectory served as the dependent variable. Participants matched the visual shape to the perceived auditory gesture, along a continuum from exponential to logarithmic curvatures, including a straight line. Only three of the four acoustic scales were compared and rated in a single trial, which allowed to assess contextual bias across all possible combinations. Preliminary data suggest that pitch glides based on the ERB rate match a straight line best, while linear-frequency glides are perceived as moderately logarithmic trajectories. These two scales, however, exhibit some contextual variability, whereas these trends seem robust across the investigated IVs, intra-subject consistency appears to strongly depend on listening expertise.

11:15

3aMU10. Correlation-based temporal model of pitch multiplicity. Peter Cariani (Hearing Res. Ctr., Boston Univ., 629 Watertown St., Newton, MA 02460, cariani@bu.edu)

Global temporal models for pitch analyze population-wide, all-order interspike interval distributions of the entire auditory nerve to accurately predict almost all known F0-pitch phenomena. Population-interval distributions (PIDs) are neural representations that resemble log-frequency scaled, half-wave rectified summary autocorrelation functions (SACF) of stimuli. PIDs produced by periodic stimuli show regular patterns of lag peaks associated with subharmonics of harmonics, with major peaks at n/F0, whereas early global temporal models chose the highest PID peak to estimate one, dominant pitch, later models compared average interval-densities of different sets of F0-related PID peaks to estimate relative pitch saliences, all saliences above a threshold being audible. However, this method inherently generates octave confusions amongst related subharmonics. Using Pearson correlation coefficients between PIDs and periodicity-related lag patterns (lags at n/F0) to estimate pitch saliences obviates the octave ambiguity problem and lag-weightings. The results from recent simulations using the Zilany-Bruce-Carney ANF model to estimate the pitches heard for dyads and triads of complex harmonic tones will be presented. For triadic chords (major, minor, suspended, augmented, and diminished), the model estimates relative saliences of F0s of notes, fundamental basses, and individual harmonics). Correlation salience values indicating relative pitch strengths (pitch stabilities) generally comport with music theoretic rankings for these chords.

11:35

3aMU11. Human-like pitch perception mechanisms in marmoset monkeys. Xindong Song, Yueqi Guo, Michael Osmanski, and Xiaoqin Wang (Biomedical Eng., Johns Hopkins Univ., 720 Rutland Ave., 412 Traylor Bldg., Baltimore, MD 21205, songxindong@jhmi.edu)

The perception of the pitch of harmonic complex sounds is a crucial function of human audition, especially in music and speech processing. Whether the underlying mechanisms of pitch perception are unique to humans, however, is unknown. Based on estimates of frequency resolution at the level of the auditory periphery, psychoacoustic studies in humans have revealed several primary features of central pitch mechanisms. It has been shown that (1) the pitch strength of a harmonic tone is dominated by resolved harmonics; (2) pitch of resolved harmonics is sensitive to the quality of spectral harmonicity; and (3) pitch of unresolved harmonics is sensitive to the salience of temporal envelope cues. Here, we show that, for a standard musical tuning fundamental frequency of 440 Hz (ISO 16), the common marmoset (*Callithrix jacchus*), a New World monkey with a hearing range similar to that of humans, exhibits all the primary features of central pitch mechanisms demonstrated in humans. Thus, marmosets and humans may share similar pitch perception mechanisms, combined with previous findings of a specialized pitch processing region in both marmoset and human auditory cortex, suggesting that these mechanisms may have emerged early in primate evolution.

Session 3aPA

Physical Acoustics, Engineering Acoustics, and Biomedical Acoustics: Acoustofluidics I

Max Denis, Cochair
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Kedar C. Chitale, Cochair
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Charles Thompson, Cochair
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Invited Papers

8:30

3aPA1. Acoustofluidics: Merging acoustics and microfluidics for biomedical applications. Tony Jun Huang (Pratt School of Eng., Duke Univ., 144 Hudson Hall, Box 90300, Durham, NC 27708, tony.huang@duke.edu)

The past two decades have witnessed an explosion in lab-on-a-chip research with applications in biology, chemistry, and medicine. Recently, a new lab-on-a-chip frontier has emerged, joining acoustics with microfluidics, termed acoustofluidics. Here, we summarize our recent progress in this exciting field and show the depth and breadth of acoustofluidic tools for biomedical applications through many unique examples, including exosome separation, cell-cell communication studies, three-dimensional bioprinting, circulating tumor cell isolation and detection, ultra-high-throughput blood cell separation, high-precision micro-flow cytometry, and portable fluid manipulation systems. These acoustofluidic technologies are capable of delivering high-precision, high-throughput, and high-efficiency cell/particle/fluid manipulation in a simple, inexpensive, cell-phone-sized device. More importantly, the acoustic power intensity and frequency used in these acoustofluidic devices are in a similar range as those used in ultrasonic imaging, which has proven to be extremely safe for health monitoring during various stages of pregnancy. As a result, these methods are extremely biocompatible, i.e., cells and other biospecimens can maintain their natural states without any adverse effects from the acoustic manipulation process. With these advantages, acoustofluidic technologies meet a crucial need for highly accurate disease diagnostics (e.g., early cancer detection and monitoring of prenatal health) and effective therapy (e.g., transfusion and immunotherapy).

9:00

3aPA2. Living probes as calibration standards for acoustic microfluidics. Minji Kim (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, St. Louis, MO), Emma Huff (Biomedical Eng., Washington Univ. in Saint Louis, St. Louis, MO), Philip Bayly, and J. Mark Meacham (Mech. Eng. & Mater. Sci., Washington Univ. in Saint Louis, 1 Brookings Dr., Urbauer Hall, Rm. 319, St. Louis, MO 63130, meachamjm@wustl.edu)

Acoustic microfluidics is promoted as an enabling technology for numerous applications in medicine and biology; however, adoption of these solutions in clinical and industrial settings is hampered by inconsistent performance and poor reproducibility. Though computational modeling and laboratory-scale demonstrations anticipate significant advantages for acoustofluidic unit operations including mixing, sorting/separation, and isolation, few technologies have realized this potential. A lack of standard tools and methods for assessing and comparing device performance represents a critical barrier to progress in the field. Here, we introduce a living probe that allows accurate and dynamically responsive measurement of the acoustic pressure within a device. Motile unicellular alga *Chlamydomonas reinhardtii* (CR) probe their environment and naturally swim against an imposed force field to fill complicated shapes. Steady-state distributions of swimming cells can be related to the field shape and strength to more completely describe the pressure field (versus passive particles that reach terminal distributions at nodal locations). Significantly, CR cells continuously respond to their environment, which enables real-time observation of the system response to varying operating conditions (e.g., frequency and/or drive voltage). We present the results that demonstrate correlation of CR cell distributions with pressure fields in simple one- and two-dimensional shapes, as well as more complex architectures.

9:30

3aPA3. Extraordinary manipulations of particles by acoustic radiation forces and torques. Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of Mississippi, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu)

Particle manipulations using radiation forces and torques produced by acoustic waves have been found valuable applications in biomedical and material engineering. Recent work suggests extraordinary manipulations of particles associated with direction reversals of forces and torques by some specific sound beams. When considering the three-dimensional acoustic radiation force, the momentum view enlightens into the superposition and coupling of manipulations by multiple beams and the force dependence on scattering functions [L. Zhang, *JASA* **144** (1), 443–447 (2018)]. It is efficient to apply the superposition to consider forces and torques on particles moved away from the equilibrium trapping position [L. Zhang, *JASA* **143**(5), 2796–2800 (2018)]. The superposition reveals how the radiation torque associated with energy absorption can spin an object around its center of mass in a direction reversed with respect to the wave vortex's handedness [L. Zhang, *Phys. Rev. Appl.* **10**(3), 034039 (2018)]. These insights into extraordinary manipulations are of interest for advancing the flexibility of particle manipulations by acoustic tweezers and to the field of acoustofluidics.

10:00–10:15 Break

10:15

3aPA4. New developments in acoustofluidics: Understanding and utilization from atomization to nanofluidics. James Friend (Mech. and Aerosp. Eng., Structural and Mech. Eng., Univ. of California, San Diego, 345F, M.S. 411 Gilman Dr., La Jolla, CA 92093, jfriend@eng.ucsd.edu)

We report new developments in understanding and utilizing acoustofluidics, including atomization devices and physics, interesting new phenomena observed in nanochannels, and entirely new directions of research. In our atomization effort, we have found an absence of bulk turbulence previously posited responsible for capillary wave phenomena that give rise to atomization and find a complex combination of caustics and nonlinear interfacial deformation to be responsible instead. These phenomena are observed on our completely portable micro-scale atomizer devices potentially useful in pulmonary applications. In extending our previous efforts in nanoslit acoustofluidics, we examine droplet manipulation in two-dimensional nanochannel arrays and observe transport at up to 0.1 m/s. Finally, we describe methods and the underpinning physics to integrate acoustofluidics into energy storage systems for a step change in their performance.

Contributed Papers

10:45

3aPA5. Calibration of an ultrasonic transducer in a standing wave field. Krishna N. Kumar, Tyler Campbell, Jack Saloio, Kedar C. Chitale (Res. & Development, FloDesign Sonics, Inc., 380 Main St., Wilbraham, MA 01095, k.kumar@fdsonics.com), and Bart Lipkens (Res. & Development, FloDesign Sonics, Inc., Springfield, MA)

Acoustofluidics is one of the emerging technologies for cell sorting. Most of the acoustofluidics platforms use an acoustic standing wave field for the separation. The calibration of the standing wave field is a challenging task. In the past, there have been few studies to quantify the standing wave indirectly using particle tracing in the field. In the present work, we attempt to calibrate the standing wave field using acoustic techniques. Hydrophones are used to find the pressure in an acoustic field. They are designed and constructed in a way that they do not alter the field. Most of the commercial hydrophone in market is made of PVDF (polyvinylidene fluoride): an acoustically transparent material. When a hydrophone is inserted in a standing wave field, the field in front of the tip is no longer a standing wave field as PVDF is acoustically transparent. This was observed in our acoustic chamber also. Here, we propose to use the receiving characteristics of a transducer to quantify the standing wave field. In brief, two identical PZT transducers are used to generate the standing wave field: one as a transmitter and other as a receiver (reflector). The receiving characteristics of the transducer used as a reflector are calibrated using a needle hydrophone. Using the signal recorded by the receiving PZT transducer in the standing wave field and the calibration value, the pressure at the surface of reflecting transducer can be found. For the sake of comparison, the field will be quantified also by the particle tracing method.

11:00

3aPA6. Theory of sound propagation through a fluidic medium with suspended clay and silt particles having the general characteristics of mud. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., P.O. Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann, and Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Work presented at previous meetings and in various publications for propagation of sound through mud continues with a theory that simultaneously takes into account viscous flow past both suspended clay and silt particles. Why silt particles do not settle under the action of gravity to the bottom of the layer is explained because (1) the clay particles adhere together in a flocculated card-house configuration and (2) the silt particles are trapped in the clay matrix. The conjecture is defended that the clay matrix and the silt particles move in lock-step under the influence of an incident sound wave. Neighboring particles of different sizes move with the same velocity amplitude, although they are subjected to different viscous forces. The theories of Stokes, Happel, and Brenner are used to calculate viscous forces at low frequencies for particles of different shapes, with the clay particles idealized as thin platelets and the silt particles idealized as spheroids with different eccentricities and random orientations. The forces on the clay particles continue to be given by the low frequency approximation for all frequencies of interest, and the deviations from Stokes's low frequency law are taken to be what corresponds to a sphere with the equivalent radius. The size distributions of the particles are taken from existing data. Modified fluid dynamic equations are derived using basic principles. The attenuation is shown to vary as frequency squared at low frequencies and as the square root of frequency for higher frequencies.

Session 3aPP

Psychological and Physiological Acoustics and Speech Communication: Context Effects in Speech Perception I

Christian Stilp, Cochair
Psychological and Brain Sciences, University of Louisville, 308 Life Sciences Building, Louisville, KY 40292

Matthew Winn, Cochair
Speech & Hearing Sciences, University of Washington, 1417 NE 42nd St., Seattle, WA 98105

Invited Papers

8:05

3aPP1. Speech perception is influenced by the speech rate of both attended and unattended sentence contexts. Hans Rutger Bosker (Max Planck Inst. for PsychoLinguist, P.O. Box 310, Nijmegen 6500 AH, The Netherlands, HansRutger.Bosker@mpi.nl)

Speech perception is influenced by the temporal properties of the surrounding acoustic context. These temporal context effects (TCEs) are contrastive: a heavily reduced target sound [tɛɹ:] is perceived as "long" "terror" when preceded by fast speech but as "short" "tear" in slow speech. I will introduce earlier studies demonstrating that TCEs involve domain-general processes arising early in perception, driven by neural oscillations entraining to the syllabic rhythm of speech. If TCEs arise early in perceptual processing, this raises the question whether TCEs are modulated by selective attention. That is, is speech perception in a "cocktail party" situation influenced by attended speech rates only or also by the speech rate of unattended talkers? In three experiments, participants were presented with two simultaneous context sentences, one in each ear, followed by diotic ambiguous target words. The speech rate of both attended and unattended talkers was found to equally influence target categorization, regardless of whether the attended context was in the same or different voice than the target, and even when participants could watch the attended talker speak. Therefore, TCEs are immune to selective attention, suggesting that TCEs largely operate at a level in the auditory processing hierarchy that precedes attentional stream segregation.

8:25

3aPP2. Distinguishing peripheral and central contributions to speech context effects. Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Acoustic context effects often function in a spectrally contrastive manner and can contribute to achieving perceptual constancy of speech sounds under widely varying conditions, including different room acoustics, different talkers, and different background noises. The neural mechanisms underlying these effects remain unknown, but they are likely to involve within-channel adaptation and across-channel inhibition, similar to that postulated for non-speech auditory context effects, such as auditory enhancement. These mechanisms could be instantiated as early as the cochlea, via efferent circuits, but may also involve higher-level processes. All such mechanisms could in principle be modulated by expectation and attention. Work from our lab has attempted to shed light on the nature and locus of the mechanisms underlying auditory and speech context effects, using a combination of techniques, including behavior, otoacoustic emissions, and electroencephalography (EEG). The results so far suggest that context effects can be somewhat modulated by attention, and may involve stages of auditory processing that extend to subcortical regions, but seem unlikely to involve changes in cochlear mechanics. [Work supported by NIH grant R01DC012262.]

8:45

3aPP3. An individual differences approach to acoustic context effects in speech categorization. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Objects and events in the sensory environment are not perceived in isolation but in the context of surrounding stimuli. This fact has been widely appreciated in the study of speech perception, as demonstrations of these context effects date back more than a half-century. For example, the spectral composition of surrounding sounds can bias categorization of later speech sounds (spectral context effects). Similarly, temporal characteristics of surrounding sounds can bias categorization of target speech sounds (temporal context effects). The literature features many individual reports of listeners experiencing these context effects, but what do these results mean across studies? If a listener exhibited a large context effect in one experiment, is that suggestive of him/her exhibiting a large context effect in a different experiment (i.e., general sensitivity to acoustic context) or not (i.e., results limited to certain stimuli / acoustic domains)? Here, I will take an individual differences approach to spectral and temporal context effects, evaluating their predictive power within and across domains, and how this might inform future work in these areas.

9:05

3aPP4. The changing social context of speech perception: Comparisons across age and time. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

Beliefs about talkers' social-group membership affect how listeners perceive their speech: listeners identify phonemes differently depending on whether they believe that the talker was a man or a woman (Strand and Johnson, 1996; Munson, 2011), whether the talker was young or old (Drager, 2011), and whether the talker is gay or straight (Mack and Munson, 2012). These social influences are thought to reflect knowledge of social norms associated with categories such as gender, age, and sexuality. Social norms themselves are subject to change. For example, attitudes toward gay and lesbian people have changed dramatically in the past 15 years. For example, a strong majority of young adults now report having a positive view of sexual minorities. This talk reports on ongoing studies in our lab that have examined whether changes in social norms about gender and sexual orientation are associated with changes in how talkers' gender and sexual orientation affect listeners' speech perception (Obeda and Munson, 2018; Obeda *et al*., 2018). We have found robust differences between archival data from our lab from 2003 and both older and younger listeners tested in 2018. This talk will discuss the implications of these findings for models of speech perception more generally.

Contributed Paper

9:25

3aPP5. Semantic context influences early speech perception: Evidence from electrophysiology. Laura Getz and Joseph C. Toscano (Psychol. and Brain Sci., Villanova Univ., 800 E. Lancaster Ave., Villanova, PA 19085, laura.getz@villanova.edu)

An unresolved issue in speech perception concerns whether top-down lexical information influences early perceptual representations. This was addressed using the event-related potential (ERP) technique to measure semantic priming effects on the auditory N1, an index of initial acoustic cue encoding. Participants saw visual primes that formed a clear association with the target (Association: "MARCHING *band*"), led to no specific association (Neutral: "BUTTER *bomb*"), or consisted of a non-word Mask. Auditory targets were stop consonants varying in voice onset time (VOT) between voiced (/b,d,g/) and voiceless (/p,t,k/) pairs. Participants were faster to identify the initial sound of the target in the Association condition than the Neutral and Mask conditions, and ERP responses showed the expected bottom-up effect of stimulus VOT (larger N1s for shorter VOTs). In Experiment 1, Association primes produced smaller N1s when targets were perfectly predictive, suggesting a top-down attentional effect. In Experiment 2, ambiguous and unexpected VOTs were added, and the results demonstrated that ambiguous VOTs in the Association condition were encoded similarly to the voicing endpoint that matched the semantic context (i.e., larger N1s for voiced expectations). These results provide the first ERP evidence that top-down lexical information directly influences early perceptual responses.

Invited Paper

9:40

3aPP6. Spoken word segmentation and contextual predictability. Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, 1290, Eugene, OR 97403, mbaesebe@uoregon.edu) and Tuuli Morrill (none, Fairfax, VA)

During speech perception, listeners rely on context when parsing the speech signal. That is, a listener's interpretation of the speech signal does not only rely on bottom-up phonetic cues in the signal. Instead, listeners use a variety of cues to interpret the best parse of a speech signal. In the current study, we examine how spoken word segmentation of potentially ambiguous stretches of speech is impacted by a variety of contextually based acoustic-phonetic cues (e.g., speaking rate) and the interactions of these cues with contextually-based linguistic knowledge (e.g., collocation strength). We additionally ask how the strength of these predictive variables varies as a function of native speaker status. That is, do non-native speakers use contextual factors in the same way that native speakers do? Our results suggest that while listeners all use both acoustic-phonetic cues and linguistic knowledge when determining the most likely parse of speech, relianceon these cues varies as a function of language status. We will discuss the implications of these results for our understanding of speech perception, generally speaking.

Contributed Paper

10:00

3aPP7. Visual primes, speech intelligibility, and South Asian speech stereotypes. Veera S. Vasandani (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, vasan007@umn.edu), Molly E. Babel (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada), and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Visual primes that suggest social attributes about a talker can affect listeners' speech perception. Recent work by Babel and Russell [*J. Acoust. Soc. Am.* **137** (2015)] showed that speech intelligibility can decrease when listeners are shown a picture of the speaker when the talker is ethnically Chinese than when the talker is not. They reason that listeners associate Chinese faces with nonnative English accents, which hinders intelligibility. This finding has important implications for our understanding of real-world speech intelligibility when talkers' and listeners' race or ethnicity differ and when these are associated with different language backgrounds. Due to ongoing demographic shifts in the US and Canada, interactions between older and younger adults are often between people with different racial, ethnic, and linguistic backgrounds. This project is a part of a larger research collaboration aimed at understanding the extent and nature of effects of race and ethnicity on speech intelligibility across the lifespan and across levels of hearing acuity. In this talk, we will discuss the results of the first of these experiments, in which we examine speech intelligibility in noise using a large set of talkers whose voices are paired with faces of individuals who are White or South Asian.

10:15–10:30 Break

10:30

3aPP8. Variability in context effects on rate adaptation within individuals. Christopher C. Heffner and Emily B. Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., U-1085, Storrs, CT 06269, christopher.heffner@uconn.edu)

Rate adaptation is a crucial aspect of speech perception. Listeners can rapidly adapt to a wide variety of speech rates. Recent evidence has indicated that these effects can emerge even over the course of single sentences: studies of what are often termed "distal context effects" have shown that the rate context at the beginning of a sentence can affect the perception of words at the end of a sentence. These effects have been found across contrasts and languages. Yet the consistency of these effects within individuals has almost never been probed. Here, we report a series of experiments that examined whether distal context effects are consistent within individual talkers. We find surprisingly low test/retest reliability in distal context effects. Namely, the strength of the effects within individuals in one session or in one set of items had almost no bearing on the strength in a second session or second set of items. This was true even when the item set was highly constrained, to mitigate item effects in the stimuli. This suggests that a more nuanced understanding of individual variation in context rate effects is necessary.

10:50

3aPP9. Development of speech recognition in noise or two-talker speech: Context effects related to response alternatives and sentence meaning. Emily Buss (UNC Chapel Hill, 170 Manning Dr., G190 Physicians, Chapel Hill, NC 27599, ebuss@med.unc.edu) and Lori Leibold (Boys Town National Res. Hospital, Omaha, NE)

Speech recognition in normal-hearing adults is affected by pragmatic restrictions on the target content. For example, masked sentence recognition is better when target speech is composed of semantically meaningful compared to anomalous sentences. Similarly, masked word recognition is better when assessed in a close-set than an open-set task, even after accounting for changes in chance performance; this effect is most pronounced when the response alternatives in the closed-set task are acoustically distinct. In both the cases, restricting the set of plausible responses reduces the fidelity of acoustic cues necessary to perform the task. Although young school-age children benefit from this type of context, the magnitude of benefit relative to that observed for adults depends on the masker type. The benefit associated with increasingly restricted response alternatives is similar for young children and adults when the masker is noise, but young children derive little or no benefit when the masker is two-talker speech. Children and adults also differ with respect to the benefit of semantic context for sentences presented in noise or two-talker speech. Potential factors responsible for these developmental effects will be discussed, including maturation of auditory stream segregation, working memory, and acoustic/phonetic templates supporting word recognition.

11:10

3aPP10. Backwards and indirect context effects in accommodating gender differences in speech. Matthew Winn (Speech-Language-Hearing Sci., Univ. Of Minnesota, 1417 NE 42nd St., Seattle, WA 98105, mwinn83@gmail.com) and Ashley Moore (Geneva Foundation, Seattle, WA)

Listeners accommodate large amount of acoustic variability across talkers and phonetic contexts when categorizing speech sounds. This study examines accommodation of gender-related talker differences, which can occur in reverse (in sound sequence AB, sound B affects perception of sound A), suggesting more complicated mechanisms than peripheral auditory contrast enhancement alone. A continuum of fricatives "sh" and "s" was preappended to vowels whose acoustic parameters were manipulated by various properties ranging from typically feminine to masculine and vice versa. Other conditions provided matching or mis-matching visual cues. We examined the weighting of cues to talker gender, the duration of exposure to sound needed to obtain the full context effect, and the influence of hearing status, using listeners with cochlear implants (CIs). CI listeners used different cues to show a comparable gender context effect and also appear to use proxy cues for acoustic properties perceived more clearly by people with normal hearing (NH). All listeners showed implicit knowledge of how acoustic patterns are selectively reverse-compatible with different articulation sequences. NH listeners needed only 30 ms for full adjustment, while CI listeners' data were much more variable. Collectively, these results show a complex process of auditory, visual and gesture-aware adjustment to a talker's voice.

11:30

3aPP11. Diagnostic precision of open- versus closed-set word recognition. Tzu-Ling J. Yu and Robert S. Schlauch (Speech-Language-Hearing Sci., Univ. of Minnesota, Twin Cities, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, yuxxx583@umn.edu)

The precision of forced-choice (closed-set) and open-ended (open-set) word recognition (WR) tasks for identifying a change in hearing was examined. WR performance for closed-set (4 and 6 choices) and open-set tasks was obtained from 70 listeners with normal hearing. Speech recognition was degraded by presenting words from the Modified Rhyme Test in speech-shaped noise (-8, -4, 0, and 4 signal-to-noise ratios) or by processing words using a sinewave vocoder (2, 4, 6, and 8 channels). The results for the two degraded listening conditions yielded similarly shaped, monotonically increasing psychometric functions. The closed-set tasks had shallower slopes and higher scores than the open-set task for the same condition. Fitted, average psychometric functions were the input to a computer simulation conducted to assess the ability of each task to identify a change in hearing. Individual data were also analyzed using 95% confidence intervals for significant changes in scores for words and phonemes. These analyses found the following for the most to least efficient condition: open-set (phoneme), open-set (word), 6-choice closed-set, and 4-choice closed-set. Greater than an order of magnitude more trials were needed for the 4-choice condition to equal the precision of the open-set word condition scored by the percentage of correct phonemes.

Session 3aSAa

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

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

Contributed Papers

8:00

3aSAa1. Propagation and scattering of Lamb waves with infinite wavelength. Clemens Gruensteidl (Mech. Eng., Univ. of Colorado at Boulder, 1111 Eng. Dr., UCB 427, Boulder, CO 80309, clgr8334@colorado.edu), David Stobbe (Lawrence Livermore National Lab., Boulder, CO), and Todd W. Murray (Mech. Eng., Univ. of Colorado at Boulder, Boulder, CO)

The dispersion characteristics of Lamb waves depend on the thickness and the elastic properties of a plate. At the cut-off frequencies (when the wavenumber goes to zero), Lamb waves are standing waves associated with plane longitudinal or shear waves reflecting up and down between the surfaces of the plate at normal incidence. In the general case, these thickness resonances have an infinite wavelength and do not carry energy along the plate. However, if the material properties are selected such that longitudinal and shear thickness mode resonances of the same symmetry coincide, then the elastic fields associated with the resonances couple to produce a wave with an infinite wavelength that propagates energy along the plate surface. Intriguing effects associated with this phenomenon can be observed in aluminum alloy plates where near-degeneracy between resonances occurs. We show that waves generated near the cutoff frequency in such cases spread from the excitation point and produce a spatially uniform oscillation over the plate surface. These infinite wavelength waves show angle independent mode conversion upon encountering the free edge of a plate or thickness step on the surface. Furthermore, we demonstrate that these peculiar waves flow around plate defects and are remarkably insensitive to scattering.

8:15

3aSAa2. Calculating the full leaky Lamb wave dispersion characteristics. Daniel A. Kiefer, Michael Ponschab, and Stefan J. Rupitsch (Chair of Sensor Technol., Univ. of Erlangen-Nuremberg, Paul-Gordan-St. 3/5, Erlangen 91052, Germany, daniel.kiefer@fau.de)

Lamb waves are mechanical waves guided in a plate with stress-free boundaries. In practice, the plate is often in contact with a fluid. The propagation of guided waves may change substantially in this case due to interaction with the adjacent medium, e.g., acoustic radiation may lead to leakage of energy into the fluid domain. The resulting plate modes are called leaky Lamb waves, and their attenuation due to leakage is an important parameter for the design of ultrasonic devices that exploit such waves. Modern methods to calculate dispersion characteristics of guided waves, i.e., the wavenumber-frequency relationship, solve the corresponding discretized eigenvalue problem. This kind of model for leaky Lamb waves with the analytically exact fluid interaction leads to a nonlinear eigenvalue problem, which is difficult to solve. We present a change in variables that yields an equivalent polynomial eigenvalue problem. The latter can be rewritten as a linear eigenvalue problem in state space. As a result, conventional numerical eigenvalue solvers can be used to robustly determine the sought wavenumbers. In contrast to traditional root-finding of the characteristic equations, the presented method guarantees to find all solutions and naturally handles complex wavenumbers. Additionally, no initial guesses for the wavenumbers are required.

8:30

3aSAa3. Temporally weighting a time varying noise field to improve Green function retrieval. Richard Weaver and John Yoritomo (Dept. of Phys., Univ. of Illinois, 1110 West Green St., Urbana, IL, r-weaver@uiuc.edu)

We consider the retrieval of Green functions G from the correlations of non-stationary non-fully diffuse noise incident on an array of sensors. Multiple schemes are proposed for optimizing the time-varying weights with which correlations may be stacked. Using noise records created by direct numerical simulation of waves in a two-dimensional multiply scattering medium, cases are shown in which conventional stacking does a poor job and for which the proposed schemes substantially improve the recovered G, rendering it more causal and/or more symmetric, and more similar to the actual G. It is found that the schemes choose weights such that the effective noise field is close to fully diffuse, with little direction dependence to its intensity.

8:45

3aSAa4. A uniform theory of diffraction formulation for acoustic radiation modes. Caleb B. Goates, Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, calebgoates@gmail.com), and Jonathan Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Acoustic radiation modes have recently regained the attention of the literature, with applications ranging from active structural acoustic control to acoustical holography. Typically the calculation of these modes requires boundary element methods for all but flat plate and full sphere geometries. This paper presents a radiation mode formulation for cylindrically curved structures with small curvature, based on the uniform theory of diffraction. This formulation shows promise, as it produces analytical expressions for the radiation resistance matrix, leading to computationally efficient radiation mode calculation. Radiation modes are shown for cylindrically curved plates, and the calculation of sound power with these modes is compared to that calculated using boundary element methods. The extension of this formulation to other curved structures is discussed.

9:00

3aSAa5. Extreme value statistics in flow-induced vibration over long time intervals. Connor McCluskey, Stephen Conlon, and Manton Guers (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804-0030, cxm1198@arl.psu.edu)

Cyclical loading on structures can lead to fatigue damage, and damage accumulation can be worsened from peak loading. In order to predict these peaks, extreme value statistics can be applied to limited vibration data. Tests must be conducted for long enough durations so that a representative population can be obtained. In extreme value statistics, the Generalized Extreme Value (GEV) distribution's parameters are also dependent on measuring a representative sample. As it is often uneconomical to test for long periods of time, the proposed experiment aims to look at the behavior of extreme value

statistics over long time records to determine if a "minimum requirement" is possible. A cantilever hydrofoil is vibrated under three flow conditions at two angles of attack. Extreme value statistics are applied to compare parameters and distributions for different record length increments. Statistics of each increment are used to generate return plots for the prediction of repeated tests. Errors are quantified to determine the accuracy of the different record lengths. The results will indicate how testing length influences GEV parameters and prediction in vibration, giving insight into duration requirements for future fatigue tests.

9:15

3aSAa6. Case study on the use of Comsol Multiphysics for undergraduate aeroacoustic research. Christopher Jasinski (Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06110, chrismjasinski@gmail.com)

The aerodynamic drag imposed by acoustic liners for commercial aircraft is a growing area of research. With the next generation of aircraft being designed for quietness and fuel-efficiency, this is expected to be a continued hot topic for the foreseeable future. There is substantial motivation to use computational techniques to evaluate the acoustic liner system, but the finite element analysis tools necessary are often inaccessible to undergraduates. Many of the mechanical engineering students at the University of Hartford learn the basics of COMSOL Multiphysics simulation software through their coursework in fluid mechanics, and this study aims to leverage this skill into more advanced research. As part of a senior capstone project, a pair of University of Hartford undergraduates will be using COMSOL to determine the acoustic impedance for various liner types and assess methods of determining the skin friction coefficient of the liner. These results will be compared to experimental work conducted over the last few years at the University of Notre Dame and NASA Langley Research Center. Experimental results, including direct drag and more detailed flow measurements, will be presented along with the computational results from COMSOL and a reflection on the ease of use for undergraduate researchers.

9:30

3aSAa7. Using measured data to enhance and extend vibro-acoustic performance predictions. Arnau Clot, Robin S. Langley (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, United Kingdom, ac2107@cam.ac.uk), Joshua Meggitt (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom), David Hawes (Dept. of Eng., Univ. of Cambridge, Cambridge, United Kingdom), Andy S. Elliott, and Andy Moorhouse (Acoust. Res. Ctr., Univ. of Salford, Salford, United Kingdom)

Modern industries usually need to ensure that their manufactured products meet certain vibro-acoustic requirements. Therefore, they have a clear need for models that can predict the broadband dynamic response of structural components at the design stage. The use of a hybrid deterministic-statistical formulation has been shown to be a suitable solution for predicting the response of a complex system in the mid-frequency range. This work explores two potential uses of experimental data to enhance and extend the applicability of the hybrid deterministic-statistical approach. Measured data are used to represent, first, complex vibration sources and, second, junctions between different structural components. The proposed uses are tested in a case study consisting of a deterministic source structure coupled to a statistical plate receiver. The approach is validated by comparing the predicted vibration response of the receiver plate to the one obtained by experimentally randomising the plate. The results show that a good agreement is obtained, both for the statistics of the receiver response and for the dynamic properties related to a point junction. It is concluded that the use of measured data can clearly extend the applicability hybrid models.

9:45–10:00 Break

10:00

3aSAa8. A high-frequency model of a circular beam with a T-shaped cross section. Andrew J. Hull and Daniel Perez (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil)

This talk derives an analytical model of a curved beam with a T-shaped cross section for use in the high frequency range, defined here as approximately 1 kHz to 50 kHz. The T-shaped cross section is composed of an outer web and an inner flange. The web in-plane motion is modeled with two-dimensional elasticity equations of motion, and the left portion and right portion of the flange are modeled separately with Timoshenko shell equations. The differential equations are solved with unknown wave propagation coefficients multiplied by Bessel and exponential spatial domain functions. These are inserted into constraint and equilibrium equations at the intersection of the web and flange and into boundary conditions at the edges of the system which produces 24 linear algebraic equations. These are solved to yield the wave propagation coefficients and this gives a corresponding solution to the displacement field in the radial and tangential directions. An example problem is formulated and compared to solutions from fully elastic finite element modeling. The accurate frequency range of this new model compares very favorably to finite element analysis up to 50 kHz. This new analytical model is over 3 magnitudes faster than the corresponding finite element models.

10:15

3aSAa9. Nonlocal homogenisation scheme for lossless and anisotropic metafluids and phononic crystals. Navid Nemati, Johann Guilleminot (Civil and Environ. Eng., Duke Univ., Hudson Hall, Office 167, Durham, NC 27708, navid.nemati@duke.edu), and Mário G. Silveirinha (Instituto Superior Técnico, Univ. of Lisbon, Lisbon, Portugal)

We present a systematic homogenisation approach to describe macroscopic dynamics of generic periodic acoustic materials composed of a rigid structure filled with a lossless fluid. This approach takes temporal dispersion as well as spatial dispersion effects into account, and can characterise, in a general manner, the oblique propagation and anisotropic effects. The theory is formulated such that all effects are encoded in, only, the effective frequency and wavevector dependent density, which can be calculated through a source-driven problem. Also, we demonstrate that the theory can homogenise materials even in frequency band gaps, including those that correspond to local-resonance phenomena or to Bragg scattering.

10:30

3aSAa10. Far-field acoustic subwavelength imaging and edge detection. Chu Ma and Nicholas X. Fang (Dept. of Mech. Eng., Massachusetts Inst. of Technol., MIT Bldg. 3-466, 77 Massachusetts Ave., Cambridge, MA 02139, machu@mit.edu)

The resolution of far-field acoustic imaging is limited due to the loss of the evanescent field that carries subwavelength information. In this work, we report on the design and experimental realization of an acoustic far-field subwavelength imaging system based on a transmitting/receiving pair for wave vector filtering and conversion. In the near-field, the transmitter is composed of a resonator array for pre-filtering the evanescent wave, and an acoustic phase grating for converting the filtered evanescent wave into propagating wave. In the far-field, a receiver that is spatially symmetrical to the transmitter is used to convert the propagating wave back to evanescent wave and perform post-filtering. By tuning the resonance frequency of the resonator array and period of the phase grating, different spatial frequency bands can be separated and projected to the far-field. Far-field imaging and edge detection of subwavelength objects are experimentally demonstrated. The proposed system brings new possibilities for far-field subwavelength wave manipulation, which can be further applied to medical imaging, nondestructive testing, and acoustic communication.

10:45

3aSAa11. Data-driven approaches for damage-type classification in vibrating square 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)

Acoustic radiation from a mechanical structure due to broadband forcing is inherently dependent on the structure's material, geometry, boundary conditions, and mechanical status. Measurements of this acoustic or vibrational response can be used to detect mechanical changes (i.e., damage) when compared to known baseline measurements of the structure. Often, however, knowledge of the *type* of damage is useful for subsequent considerations. Even for relatively simple structures, like flat plates, the lack of simple

solutions to problems involving localized damage makes analytical treatments challenging. This is further complicated since the location and severity of damage are typically unknown a priori. We present a data-driven approach for classification of various forms of damage—including cuts, localized mass changes, delamination, and boundary changes—in a vibrating clamped square plate. Frequency response curves are generated using FEA of a 30-by-30-by-0.3-cm aluminum plate with various types and severity of damage. Features (including changes in peak-locations, -amplitudes, and -widths compared to baseline) are extracted and used to train classifiers, with performance quantified using auxiliary test data. Comparisons are made between classifier types, including nearest neighbor methods, discriminant analysis, and support vector machines. [Work supported by NAVSEA through the NEEC and the US DoD through an NDSEG Fellowship].

WEDNESDAY MORNING, 15 MAY 2019 SEGELL, 10:45 A.M. TO 12:00 NOON

Session 3aSAb

Structural Acoustics and Vibration, Engineering Acoustics, and Noise: Noise and Vibration in Rotating Machinery

Robert M. Koch, Cochair
Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708

Elizabeth A. Magliula, Cochair
Division Newport, Naval Undersea Warfare Center, 1176 Howell Street, Bldg 1302, Newport, RI 02841

Chair's Introduction—10:45

Invited Papers

10:50

3aSAb1. Decreasing the radiated acoustic and vibration noise of autonomous underwater vehicles. Gerald L. D'Spain, Richard Zimmerman, Dennis Rimington, and Scott A. Jenkins (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu)

An outstanding challenge in ocean engineering is decreasing the radiated acoustic and vibration noise of autonomous underwater vehicles (AUVs) while transiting at high speed. At low speed below 4–5 kt, our previously published results for both a modified mid-size, prop-driven AUV and a large (30 liter) buoyancy driven glider demonstrate that, with proper engineering, own-platform-created noise levels recorded by hull-mounted hydrophone arrays can be decreased below typical background ocean noise levels across the frequency band above a few hundred hertz. Below this frequency, the remaining noise is primarily vibration induced. Recent efforts have focused on decreasing propulsion noise at high speed from prop-driven platforms. Since buoyancy can be changed only over a very short portion of the dive cycle, buoyancy-driven propulsion systems provide an alternative approach to achieving low-self noise, high-speed, underwater flight. However, vibration and flow noise created by hydrodynamic forces still can present challenges at low frequencies. [Work supported by the Office of Naval Research and BP.]

11:10

3aSAb2. Perception of complex tones in noise. Jan Hots, Gloria-Tabea Badel, and Jesko L. Verhey (Dept. of Experimental Audiol., Otto von Guericke Univ. Magdeburg, Leipziger Straße 44, Magdeburg 39120, Germany, jan.hots@med.ovgu.de)

We are often surrounded by technical sounds that contain tonal components, which usually are generated by rotating parts of machinery. Tonal components commonly reduce the pleasantness of a sound, i.e., these sounds are perceived as less pleasant compared to equal-level sounds without tonal components. To consider the influence of tonal components on noise pollution, different standards include sections dedicated to this effect. For single tonal components, previous data showed that tone adjustments calculated on the basis of the German standard 45681 account quite well for the reduced pleasantness. However, the standard is limited to single tonal components only. It was also shown that the partial loudness of the tonal portion corresponds to the perceived magnitude of tonal content (also known as tonalness or tonality). The present study investigated the perception of complex tones in noise. To this end, the loudness of the tonal portion of the sound is determined for sounds containing a single tone or a complex tone with two or four components at different levels above the masked threshold. It is shown that an increase in components results in an increase in the magnitude of the tonal content. This likely also affects the pleasantness of the sound.

11:30

3aSAb3. Quadcopter propeller noise and its control. Timothy A. Brungart, Stephen T. Olson, Brian L. Kline, and Zachary W. Yoas (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804-0030, tab7@arl.psu.edu)

Simple scaling analysis is used to show that truly significant reductions in propeller noise are possibly by increasing the propeller diameter while reducing its rotational speed, thereby reducing its blade tip speed, while maintaining a given level of static thrust. The scaling analysis is also used to show that efficiency improvements accompany the reductions in radiated noise. The significant reductions in noise and the increase in efficiency predicted were verified experimentally by measuring the radiated noise and power requirements for both large, slowly rotating propellers and small, high speed propellers, at equivalent static thrusts, where the large propellers serve as possible replacements for the small propellers in typical quadcopter applications. The bulk of the noise reductions offered by large and slowly rotating propellers compared with small, high speed propellers is shown to be maintained at equivalent net force (thrust minus weight conditions), as required for practical implementation into a quadcopter, even using readily available hobby grade components. Further significant noise reductions are possible with lightweight and custom engineered components.

11:45

3aSAb4. Nonlinear response of a flanged fastened joint due to acoustical sine sweep excitation. Trevor W. Jerome, Micah R. Shepherd, and Stephen Hambric (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, MS 3220B, State College, PA 16804-0030, twjerome@gmail.com)

Fastened joints can significantly complicate the prediction of damping and resonance frequencies of built-up systems. This complexity often leads to over-designing for safety, which drives up manufacturing and operational costs. To improve understanding of the dynamic response of fastened joints, impact hammer and acoustically driven excitation of fastened plates at a flange have been measured. These findings add experimental data that are not well represented in the literature for such joints. The structure was excited acoustically, and acceleration was measured at multiple locations on the flange and plate. Backbone curves were generated with sinusoidal sweeps of acoustic input energy. These curves highlight the nonlinearity of the system, which varies according to fastener preload. Generally, as preload torque values decrease, nonlinearity increases with shifts observed in damping, stiffness, and resonance frequency. Data presented will be used to create a model for better predicting the global dynamic properties of built-up structures.

WEDNESDAY MORNING, 15 MAY 2019 GRAND BALLROOM C, 9:00 A.M. TO 12:00 NOON

Session 3aSC

Speech Communication: Developmental and Clinical Populations (Poster Session)

Terrin N. Tamati, Chair
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All posters will be on display, and all contributors will be at their posters from 9:00 a.m. to 12:00 noon. To give contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

3aSC1. A real-time, automated tongue tracking method for ultrasound biofeedback in speech therapy. Jack A. Masterson, Sarah R. Li, Hannah M. Woeste (Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, ML 0586, Cincinnati, OH 45267-0586, masterjk@mail.uc.edu), Sarah Dugan (Commun. Sci. and Disord., Univ. of Cincinnati, Dayton, OH), Neeraja Mahalingam (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Suzanne E. Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Michael A. Riley (Psych., Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Midsagittal ultrasound images of the tongue surface are currently used as feedback on tongue motion for speech therapy, but can be difficult to interpret. Ultrasound biofeedback would benefit from a simplified, immediate representation of tongue motion. Tongue tracking in ultrasound is complicated by reflections from structures near the tongue surface, complex tongue shapes, and rapid, extreme tongue movements. Available programs that trace contours of the tongue surface have not been implemented in real time and can require extensive user input. We offer a tongue tracking method that can operate in real time and requires minimal user input. This method applies a low-pass filter to the B-mode image and then maps the tongue surface based on local brightness maxima within search regions estimated by 2nd-order Taylor series expansions in space and time. Required user input includes a graphically selected calibration point near the tongue surface and relative brightness thresholds for anterior and posterior ends of the tracked surface. Preliminary results show that this method can capture complex, rapid movements of tongue parts during production of American English /r/. Efficient tongue tracking enables real-time analysis of tongue dynamics, offering a valuable tool for speech therapy.

3aSC2. An ultrasound study of American English laterals produced by first graders. Sherman D. Charles and Steven M. Lulich (Speech & Hearing Sci. and Linguist, Indiana Univ., 1610 S Dorchester Dr. Apt. 49, Bloomington, IN 47401, sdcharle@indiana.edu)

Longitudinal studies of child speech production development have traditionally relied on perceptual evaluation and acoustics to track acquisition norms and phonological patterns. The present study administers novel techniques in speech imaging to investigate the articulation and acoustics of speech sounds in developing children, focusing on the lateral sound /l/ as produced by first graders. Real-time three-dimensional ultrasound images with synchronous audio and video recordings were collected from L1 English speaking children. The stimuli were sourced from the GFTA-3—a clinical picture-naming instrument—which was administered by a speech-language pathology student-clinician. Three-dimensional tongue surface reconstructions and the accompanying audio signals were then compared across tokens and across speakers to find common and divergent patterns for lateral sounds. The results show highly variable articulatory strategies that nevertheless can be categorized into three groups. The wide articulatory variety is not obviously tied to acoustic variability, which is relatively small; formant positions are relatively consistent across tokens and across subjects, with few outliers. From the acoustics alone, these children consistently produce what is traditionally described as the dark /l/, but articulation suggests greater variety.

3aSC3. Dyslexia impedes the processing of vocoded speech. Robert A. Fox, Ewa Jacewicz, Gayle Long, and Zane Smith (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

Research has shown that phonological impairment in dyslexia (DYS) is associated with a deficit in recognizing fine-grained details in voices of multiple talkers in both clear and degraded speech [Fox *et al*., *J. Acous. Soc. Am.* **141**, 3839 (2017)]. These results suggest that DYS listeners may have difficulties in the processing of vocoded speech due to the absence of harmonic information. Given the extensive variability in speech including gender and dialect of the speakers, we can predict that DYS individuals may have limitations using cochlear implants. In the current study, 20 DYS adults and 20 corresponding control listeners heard a set of short sentences/phrases produced by male and female talkers from Central Ohio or Western North Carolina. The utterances were unprocessed or processed through a noise-source vocoder at 4, 8, 12, and 16 channels. In separate experiments, listeners were asked to identify the dialect of the speaker (an ID task) or to repeat the sentence/phrase heard (an intelligibility task). The ID responses (A' scores) of the DYS listeners were lower than those of the controls at every stimulus level. Similarly, intelligibility responses demonstrated that DYS listeners performed more poorly than did control listeners at every stimulus level.

3aSC4. Implant location in type 1 thyroplasty for unilateral vocal fold paralysis: Measurements from excised canine larynges. Alexandra Maddox, Charles Farbos de Luzan, Liran Oren, Sid M. Khosla, and Ephraim Gutmark (Univ. of Cincinnati, 1118 Broadway St., Cincinnati, OH 45202, maddoxa219@gmail.com)

Patients with unilateral vocal fold paralysis (UVFP) complain of a soft breathy voice that is hard to understand. Thyroplasty Type 1 (TT1), the most common surgical intervention for UVFP, uses an implant to push over the membranous fold. There remains controversy regarding TT1 procedures. Questions include the optimal shape and size for the implant, where to place the implant and whether an arytenoid adduction should be added. Despite several studies, no optimal technique has been found, and the revision rate has reported to be as high as 12%–25%. This study aims to understand the impact of implant location on vocal fold vibration. TT1 procedures were performed on excised canine larynges using silastica implants placed either at the glottis or just below the glottis to medialize the folds. Larynges were then made to phonate at various subglottal pressures (Psg), and measurements were taken of the acoustics, supplied flow rate, and Psg. From these measurements, the glottal efficiency was calculated, and the acoustic signal was analyzed to determine the quality of the sound source. On average, larnges medialized in the subglottis had higher glottal efficiency and cepstral peak prominence than those medialized at the glottis, indicating that lower implant location is preferable.

3aSC5. Dysarthric speech perception: Comparison of training effects on human listeners versus automatic speech recognition tools. Michael F. Lally (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 N Wright St., Urbana, IL 61801, mlally2@illinois.edu), Heejin Kim (Linguist, Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Lori A. Moon (Psych., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Dysarthric speech, a motor speech disorder associated with neuro-motor conditions, poses a challenge for automatic speech recognition (ASR) due to its acoustic characteristics and the comparatively limited volume of available data. This study investigates ASR toolkits' recognition abilities on speech from speakers affected with cerebral palsy (CP) at different intelligibility levels, by comparing them against human listeners' performance. We ask: (1) how intelligible is speech from CP-affected speakers to ASR toolkits trained on non-dysarthric speech? (2) how does this performance compare to naïve human listeners? and (3) as familiarized human listeners better understand dysarthric speech, to what degree is dysarthric speech more intelligible to similarly familiarized ASR toolkits? Using the UA-Speech Database (Kim *et al*., 2008), we test the ASR system with two training methods: strong supervision, with both audio and orthography feedback in training, and unsupervised methods, with only audio signals, following Kim and Nanney's (2014) experiments with human listeners. ASR accuracy is measured by the word error rate in word transcription tests. Findings reveal the extent to which supervision affects ASR models in comparison to human listeners. Implications regarding how to improve the adaptation techniques to dysarthric speech for both ASR and human listeners are presented.

3aSC6. Mandarin Chinese consonant production of patients with cleft palate. Heng Yin, Xi Wang, Chunli Guo, Jingtao LI, Yang Li, and Bing Shi (Dept. of Cleft Lip and Palate Surgery, West China College of Stomatology, West China Hospital of Stomatology, Sichuan Univ., Chengdu, Sichuan 610041, China, yinheng@scu.edu.cn)

Eighteen Mandarin Chinese consonants were recorded from 259 patients with cleft palate in a /CV/ context. The age of participants ranged from 8 to 40 years old. Participants had five types of cleft palate: submucous cleft palate, hard and soft cleft palate, unilateral complete cleft palate, bilateral complete cleft palate, and congenital velopharyngeal insufficiency. All consonants were transcribed by speech-language pathologists. Results showed that the accuracy of consonant production was significantly affected by the type of cleft palate, consonant category, and their interaction. The correctness of consonant production ranged from 45.8% for the type IV cleft palate (e.g., BCCP) to 71.5% for Submucous cleft palate. In particular, for the manner of articulation, nasals had significantly higher accuracy than stops, fricatives, and affricates, while for the place of articulation, palatal and labial sounds were problematic with low accuracy. In addition, the errors of consonant production, such as replacement, compensation, and omission, were also dependent on the type of cleft palate and consonant category and their interaction.

3aSC7. Efficacy of two speech-perception training methods for aided listeners with mild-to-moderately-severe age-related hearing losses. James D. Miller, Charles S. Watson, and Gary R. Kidd (Res., Commun. Disord. Technol., Inc., 3100 John Hinkle Pl, Ste 107, Bloomington, IN 47408, jamdmill@indiana.edu)

Preliminary analyses of interrelations among variables that correlate with measures of speech perception by aided listeners with mild-to-moderately-severe age-related hearing loses are described. The SPATS Group of nearly 120 hearing-aid users was trained in quiet and noise to identify syllable-constituents and to identify words in simple sentences. The Listen Group of about 60 aided listeners listened for an equal amount of time to recorded narratives. All were given audiometric, working memory, intelligence tests, a battery of psycho-acoustic tests, and a battery of speech-perception tests prior to training. The audiometric and speech perception tests were repeated after training and again after a 3-month retention period. Prior to training, there were large individual differences speech perception scores

that could only be partially accounted for by the severity of audiometric loss. Training appears to be more effective for the SPATS Group than for the Listen Group. While improvements with training were generally modest, those who were very good or very poor initially tended to show little or no improvement, while those with middling scores tended to show more improvement. Sentence scores are highly correlated with syllable-constituent scores and with the use of context, and the use of context is correlated with working memory. (Miller and Watson are stockholders in communication Disorders Technology, Inc., and may profit from sales of some the software used in this study.)

3aSC8. Contributions to diminished perceptual learning in individuals with language impairment. Nikole Giovannone and Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT, nikole.giovannone@uconn.edu)

The lexical context in which sounds occur can help listeners mitigate the challenges of the lack of invariance in the speech signal. Through lexically guided perceptual learning, phonetic category structure can be dynamically modified given repeated exposure to potentially ambiguous sounds embedded within familiar lexical contexts. Individuals with language impairment (LI) show deficits in general auditory processing and speech perception, which may influence the development of stable phonetic category structure. However, it is not yet known whether deficits in phonetic category structure in individuals with LI are related to impairment in the learning mechanisms that guide adaptation or in feedback links from the lexicon to speech sounds. To assess these sources of deficit, participants will complete measures of receptive language ability, a task assessing lexical recruitment (i.e., the Ganong effect), and a lexically guided perceptual learning task. If impairment in phonetic category structure stems from weaker top-down lexical influences, then performance on the lexical recruitment task should be (1) diminished in individuals with LI and (2) predictive of lexically guided perceptual learning. If diminished adaptation instead reflects a disrupted learning mechanism, then individuals with LI and typical participants should perform comparably on the lexical recruitment task despite showing differences in lexically guided perceptual learning.

3aSC9. The effect of talker variability on word recognition under cochlear implant simulation. Terrin N. Tamati and Deniz Baskent (Dept. of Otorhinolaryngology / Head and Neck Surgery, Univ. Medical Ctr. Groningen, Hanzeplein 1, Groningen 9700RB, The Netherlands, t.n.tamati@umcg.nl)

Normal-hearing (NH) listeners have been shown to be less accurate and/or slower to recognize spoken words when the talker changes from trial to trial, compared to when the talker remains the same. Less is known about the effect of talker variability on cochlear implant (CI) users, who display a limitation in talker perception. As a first step, the current study investigated how limited talker information via CI simulation influences word recognition. In two experiments, NH listeners completed a word naming task in different talker conditions (single (ST), multiple female (MT-F), and multiple female/male (MT-M)) and simulation conditions (12, 8, and 4 vocoder channels). An effect of talker variability was observed, but only when talker differences were maximized. Listeners were less accurate, but not slower, at recognizing words under the MT-M condition, compared to ST and MT-F conditions. Talker variability effects did not vary by the simulation condition. These results are consistent with previous studies with NH listeners, showing similar performance in conditions where talker changes were not detected but talker differences were large. Taken together, these findings suggest that limitations in talker perception for NH listeners under CI simulation, and potentially CI users, alters the perceptual strategy for recognizing words under multiple-talker conditions.

3aSC10. Flow characteristics as a function of velopharyngeal gap size. Michael Rollins (Biomedical Eng., Univ. of Cincinnati, MSB 6303, 231 Albert Sabin Way, Cincinnati, OH 45267, rollinmk@mail.uc.edu) and Liran Oren (Otolaryngol., Univ. of Cincinnati, Cincinnati, OH)

The classic paradigm for treatment of velopharyngeal insufficiency (VPI) is based on the assumption that the severity of the speech disorder correlates with the size of the velopharyngeal (VP) opening. However, several studies have documented that even small VP openings can cause significant speech distortion. To investigate the potential contribution of aeroacoustic sources to speech distortion in VPI-affected speech, we set out to determine how airflow characteristics change as a function of the VP gap size. To do so, a head and neck CT scan of a VPI patient sustaining the phoneme /z/ was obtained. From the CT scan, patient-specific phantoms of the upper airway were made extending from the glottis to the nasal and oral exits. Four phantoms were made: one with the original VP gap size and three with VP gap sizes artificially scaled by factors of 1.25, 1.5, and 2. Particle image velocimetry (PIV) was used to measure the velocity and turbulence fields along planar slices in each phantom. Average velocity fields for each VP gap size are presented, and the change in flow characteristics as a function of the VP gap size is discussed, along with the implications for aeroacoustic sources.

3aSC11. Detecting changes in lingual differentiation over the course of speech sound intervention. Heather M. Kabakoff, Graham T. Feeny (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, heather.kabakoff@nyu.edu), Mark Tiede (Haskins Labs., New Haven, CT), Jonathan L. Preston (Commun. Sci. and Disord., Syracuse Univ., Syracuse, NY), and Tara McAllister (Communicative Sci. and Disord., New York Univ., New York, NY)

"Lingual differentiation" describes the ability to articulate different regions of the tongue semi-independently. Children with speech sound disorder (SSD) exhibit fewer differentiated gestures than typically developing children (Gibbon, 1999) and show a greater degree of lingual differentiation after treatment than before treatment (Preston *et al*., 2018). In the current study, four preschool-aged children (4;1-5;4) were treated for error patterns identified on the HAPP-3 measure in 18 sessions of cycles treatment (Hodson and Paden, 1983) over six weeks. Ultrasound and audio recordings were collected as children named pictures containing a variety of target sounds at pre-treatment, during six within-treatment sessions, and at post-treatment. For each production, lingual contours were extracted from midsagittal ultrasound images and quantified using two complexity metrics: modified curvature index (Dawson *et al*., 2016) and number of inflections (Preston *et al*., 2018). Trained listeners transcribed each production and calculated the percentage consonants correct (PCC) at each time point using Phon speech analysis software (Hedlund and Rose, 2016). We ask whether incremental changes in lingual differentiation are detectable in preschool-aged children enrolled in treatment, and whether these changes correlate with the increase in production accuracy (PCC). We also examine gradient differences in lingual complexity for rhotic tokens transcribed as misarticulated.

3aSC12. Effect of number of channels and speech coding strategy on speech recognition in mid-scala electrode recipients. Katelyn A. Berg (Dept. of Speech & Hearing, Vanderbilt Univ., Ste. 8310, 1215 21st Ave. South, Nashville, TN 37232, katelyn.a.berg@vanderbilt.edu), Jack H. Noble, Benoit Dawant (Dept. of Elec. Eng. & Comput. Sci., Vanderbilt Univ., Nashville, TN), Robert Dwyer (Dept. of Speech & Hearing, Vanderbilt Univ., Nashville, TN), Robert Labadie (Dept. of Otolaryngol., Vanderbilt Univ., Nashville, TN), and René H. Gifford (Dept. of Speech & Hearing, Vanderbilt Univ., Nashville, TN)

This study investigated the number of channels needed for maximum speech understanding and sound quality in 5 (anticipated n = 10) adult cochlear implant (CI) recipients with mid-scala electrode arrays completely within scala tympani (ST). CI programs with 4–16 active electrodes using a CIS-based strategy were created along with two n-of-m programs (8-of-16 and 12-of-16). Measures of speech understanding and sound quality were assessed. Our hypothesis was that individuals with precurved electrodes localized in ST would have greater channel independence than previous studies demonstrating gains in performance beyond 8 channels. Participants demonstrated performance gains up to 8–10 electrodes for speech recognition and sound quality ratings. Significantly poorer performance was observed using an n-of-m strategy as compared to CIS conditions with 8+ electrodes. These data are in contrast with recent studies for precurved arrays (e.g., Croghan *et al*., 2017; Berg *et al*., in review) reporting significant improvement up to 16 to 22 channels. However, the current results are

consistent with previous literature (e.g., Fishman *et al*., 1997; Friesen *et al*., 2001; Shannon *et al*., 2011), demonstrating no more than 8–10 independent channels for CI recipients with straight arrays. Implications for device selection and the impact of electrode-to-modiolus distance will be discussed.

3aSC13. Acoustic measures of paraspeech in speakers with multiple sclerosis: A study of neuropsychological status and its role in diadochokinesis. Lynda Feenaughty (Univ. of Memphis, 4055 N Park Loop, Memphis, TN 38152, lynda.feenaughty@memphis.edu)

Paraspeech tasks are frequently incorporated in motor speech disorder assessments and include syllable diadochokinesis (DDK), the rapid repetition of alternating movements (Kent, 2015). DDK measures indicate movements of the oral articulators and may detect problems before specific functions such as speech are affected in neurodegenerative diseases such as multiple sclerosis (MS). Although there is growing appreciation from studies of various clinical populations that cognitive and speech motor processes influence or interact with one another (Kent, 2004), the impact of cognitive limitations of speakers with dysarthria on DDK performance is not well understood. Toward this end, the current study explored cognitive status as a factor in DDK performance for 48 speakers with MS and 12 healthy adults. All speakers were recorded as they repeated various syllables as quickly and steadily as possible. Standard acoustic criteria were used to obtain global and segmental temporal measures of DDK using speech analysis software. In addition, all speakers underwent rigorous cognitive testing (e.g., information processing efficiency). Because controversy continues over the use of DDK, this study sought to provide evidence for the clinical value of paraspeech in the assessment of motor speech disorders as well as cognitive status. [Work supported by the ASH Foundation and the University at Buffalo MDRF.]

3aSC14. Masking effects of perceptual cues on hearing-impaired ears. Siyan Guo, Cliston Cole, and Jont Allen (ECE, Univ. of Illinois, Urbana-Champaign, 306 N Wright St., Urbana, IL 61801, sguo16@illinois.edu)

The most common complaint from hearing-impaired (HI) subjects is I can hear the speech but I can't understand it. This has led us to investigate the strategy of the HI ear in understanding speech. In this study, the effects of the masking of primary cues in HI ears are analyzed, with and without the presence of conflicting cues. Two consonant-vowel (CV) identification experiments were conducted on 5 normal-hearing (NH) and 10 HI subjects. 4 plosive consonants /t,k,d,g/ paired with vowel /a/ were used as target stimuli. The CVs were presented at SNRs of 0, 9, and 18 dB. In experiment I, the primary cue for each CV was modified in 3 ways: removed, attenuated, and amplified. Experiment II was similar except the conflicting cues were removed. The results for both NH and HI groups show a large increase in probability of error in all SNR levels when the primary cue was removed. The results for Exp II display more variability. The HI subjects were divided into three groups based on their error: low error group (LEG), medium error group (MEG) and high error group (HEG). The analysis shows that HI listeners are using the same primary cue as NH ears. The strength of the primary cue is a critical quality for correct speech perception, especially in the presence of noise. The subjects in MEG and HEG exhibit sensitivity to the presence of conflicting cues. The removal of the conflicting cues demonstrates that HI ears use not only the primary cue but also conflicting cues, for correct speech perception.

3aSC15. Whole word scoring versus phoneme error scoring for audiological word recognition testing. Edward L. Goshorn, Jennifer Goshorn, and Jennifer Arnoult (Speech and Hearing Sci., PsychoAcoust. Res. Lab., Univ. of Southern Mississippi, 118 College Dr. #5092, Hattiesburg, MS 39401, edward.goshorn@usm.edu)

Conventional audiological word recognition testing uses whole word scoring for errant responses resulting in a percent correct score. This approach is found to be lacking in diagnostic utility. Recently, phoneme error scoring was suggested as a supplement to whole word scoring in attempt to improve diagnostic utility but is also limited in that it is highly correlated with the whole word score. This project examined the use of phoneme error scoring that applies an exponential equation to the type and the number of phoneme errors yielding a score that may be compared to absolute hearing sensitivity loss as measured by conventional audiograms. Because the number of words used in audiological word recognition varies, a correction for the number of stimuli is applied. Archival data, case study data, and simulated cases were used to evaluate and compare the diagnostic utility of three scoring approaches for errant responses: whole word, total phoneme errors, and an exponential phoneme error score (EPES) that weights consonant and vowel errors differently: EPES = number of phoneme errors times 2^{number} of consonant errors + 10^{number} of vowel errors. The case study and simulated results were used to evaluate the utility of each scoring approach.

3aSC16. Interactions among perceived breathiness, roughness, and overall severity of dysphonia. David A. Eddins, Supraja Anand (Commun. Sci. & Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620, deddins@usf.edu), and Rahul Shrivastav (Vice President for Instruction, Univ. of Georgia, Athens, GA)

Dysphonic voices typically co-vary across multiple quality dimensions. This study investigates the interaction between breathiness and roughness and their combined contributions to judgments of overall dysphonia severity. Four dysphonic talkers were replicated using a Klatt synthesizer with the Liljencrants-Fant (LF) model as the glottal excitation source. Two vowel continua were created for each speaker by systematically varying aspiration noise (AH) and open quotient (OQ) [to vary the magnitude of breathiness] and the waveform amplitude modulation depth [to alter the magnitude of roughness]. The stimulus matrix for each talker comprises 49 stimuli (7 breathy levels X7 rough levels) resulting in a total of 196 stimuli (4 talkers X49 stimuli).Ten naïve listeners provided judgments of breathiness, roughness, and overall dysphonia severity using a magnitude estimation task. Analyses determined the interaction between talker and listener judgments of breathiness, roughness, and overall severity. A set of iso-severity curves from the ME task were derived and illustrated the combination of breathiness and roughness magnitude to dysphonia severity. These data will be useful in establishing the validity of clinical scales for voice quality perception. [Work supported by NIH DC009029.]

3aSC17. Differential impact on speech production from chronic thalamic DBS in Essential Tremor and that from STN DBS in Parkinson's disease. Emily Wang (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Ste. 1017 AAC, 600 South Paulina, Chicago, IL 60612, emily_wang@rush.edu) and Leonard A. Verhagen Metman (Neurological Sci., Rush Univ. Medical Ctr., Chicago, IL)

Essential Tremor (ET) is the most common tremor syndrome seen in adults. The characteristic tremor in ET is postural and action tremors, with a frequency of 4–7 Hz; while in patients with Parkinson's disease (PD), the dominant tremor is resting tremor with a typical frequency of 5 Hz. Deep brain stimulation (DBS) in the ventral intermediate nuclei (Vim) has been shown effective in treating action or intention tremors in medically resistant ET, while DBS in the subthalamic nucleus (STN) has been shown effective in treating rigidity, rest tremor and Dopa-induced dyskinesias in PD. However, chronic bilateral DBS, whether in Vim or STN, may have negative impact on speech production. We report two cases where both patients received DBS greater than one year and developed speech impairment. When the stimulation in Vim was turned off for 30 min in ET, the patient's speech returned to the baseline while turning the STN stimulation off for 30 min in PD, minimal changes were observed and 12 h with stimulation-off were required for speech to return to the non-stimulated state.

3aSC18. Variations of F2 range measures in mild dysarthria and healthy talkers. Michaela McLaughlin and Lynda Feenaughty (School of Commun. Sci. and Disord., The Univ. of Memphis, 4055 N Park Loop, Memphis, TN 38152, mmclghl2@memphis.edu)

Diminished vocalic segments indexed by the extent, duration, and rate of change in the second formant (F2) are common in dysarthria and are known to reduce intelligibility in adults with multiple sclerosis (MS; Hartelius *et al*., 2010). When F2 range was measured over select sentences from a reading passage, results indicated that the F2 range may not be sensitive to

mild dysarthria in MS (Rosen *et al*., 2008). However, measures of the F2 range should be further explored for sentences imposing greater speech-motor and cognitive-linguistic demands. As such, phonetic contexts requiring even small vocal tract movement may contribute to intelligibility for speakers with mild dysarthria in MS. Thus, this study will investigate the impact of sentence length and complexity on within-speaker variations of F2 movement and group differences for 12 adults with mild dysarthria in MS and 12 healthy talkers. Various F2 range measures shown to be sensitive to speech motor capabilities will be obtained from 11 audio-recorded sentences varying in length and complexity. The results of this study will provide a better understanding of the speech motor control capabilities of speakers with mild dysarthria with relatively preserved intelligibility in MS. [Work supported by ASH Foundation and the University at Buffalo MDRF.]

3aSC19. Response times of repeated nonwords in children with residual speech sound disorder. Caroline E. Spencer, Sisan Cuervo, Natalie Baldinelli, Caroline Sheehan, and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., P.O. Box 670379, Cincinnati, OH 45267, spenceco@mail.uc.edu)

Nonword repetition (NWR) is sensitive to impairments in a range of disorders, including speech sound disorders (Larrivee and Catts, 1999). NWR requires listeners to perceive an auditory stimulus, store the perception in working memory, and repeat the stimulus aloud when prompted. Response time—from stimulus presentation to participant response—can be measured to assess processing speed. Deficits in each of these processes have been linked to speech sound errors in young children (Bird, *et al*., 1995; Lewis, *et al*., 2011; Preston, *et al*., 2015; Rvachew, *et al*., 2003), but research from older children with *residual* speech sound disorders (RSSD) is lacking. To evaluate processing speed in children with RSSD, we have adapted a nonsense Syllable Repetition Task (SRT; Shriberg *et al*., 2009). The SRT consists of multisyllabic nonsense words that use targeted phonemes (/b, d, m, n, ɑ/), to minimize the possibility of articulation difficulties masking the skills intended to be studied—phonological processing. Response times on 2-, 3-, and 4-syllable nonwords will be compared between children with RSSD and children with typically developing speech. Based on preliminary findings, we suspect that response times of children with RSSD are slower than those of children with typical speech. The results will be discussed.

3aSC20. Conversational turn-taking in youth at clinical high-risk for psychosis. Laura Sichlinger (King's College London, London, United Kingdom), Emily Cibelli, Matthew Goldrick, and Vijay Mittal (Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 608, matt-goldrick@northwestern.edu)

Previous studies of speech abnormalities in schizophrenia reflect a consensus that atypical turn-taking behavior during conversation is a prominent feature of speech patterns in psychosis. However, the literature focuses on the population with formal psychosis and lacks studies that involve individuals who are at clinical high-risk (CHR) for developing a psychotic disorder. This study assessed the turn-taking performance of CHR youth to evaluate if abnormalities in turn-taking can be observed in the CHR population compared with controls and to investigate if turn-taking performance is correlated with symptom severity. While analysis of speech data from structured clinical interviews shows no significant group effects, high scores in attenuated psychosis symptoms were associated with significantly longer between turn pauses in the CHR population. This study adds to a growing body of literature that suggests that patterns of speech and linguistic features may be useful in detecting risk, and tracking clinical and treatment course across schizophrenia and spectrum disorders.

3aSC21. Predicting children's word recognition accuracy with two distance metrics. Emma Brown (Dept. of Speech and Hearing Sci., Indiana Univ., 0 S. Jordan Ave., Bloomington, IN 47405, emebrown@indiana.edu), Izabela A. Jamsek, Laura Liang, Rachael F. Holt (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH), and Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Children generally have more difficulty in recognizing words produced by talkers with unfamiliar regional dialects and nonnative accents compared to their home dialect, but the specific deviations from native norms that cause these word recognition decrements have not been quantified. This study examines the relation between word recognition accuracy and two distance metrics: phonemic pronunciation distance and holistic perceptual distance. To calculate phonemic pronunciation distances, sentences from six speakers representing different regional dialects (British and Scottish English) and nonnative accents (Japanese-, German-, Mandarin-, and Hindi-accented English) were phonemically transcribed and compared to transcriptions from speakers of the ambient dialect (Midland American English). Preliminary results show a relation between the phonemic pronunciation distances and children's word recognition accuracy for the sentences. Follow-up work will measure holistic perceptual distances through a ladder task in which listeners will rank a larger set of native and nonnative talkers based on the perceived distance to the home dialect, a measure that will capture both segmental and suprasegmental differences. Comparing the two distance metrics will provide mechanistic insight into the impact of extent and type of talker deviation from native norms on children's word recognition accuracy. [Work supported by National Science Foundation #17570 and Indiana University Hutton Honors College.]

3aSC22. The (minimal) influence of perceived age on the perceived gender of children's speech. Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

The acoustic characteristics of boys and girls' speech resemble adult norms for men and women, respectively, early in life. Moreover, the extent to which a child's speech resembles the adult norms for their biological sex is related to other measures of the extent to which their gender development meets cultural norms [e.g., Perry *et al*., *J. Acoust. Soc. Am.* **109** (01); Munson *et al*., *J. Acoust. Soc. Am.* **137** (15); and Beaulieu *et al*., *J. Acoust. Soc. Am.* **143** (18)]. The acoustic cues to gender in children's speech are distributed throughout the speech signal. Hence, the most common method to determine the attainment of gendered speech is by gathering perceptual ratings of gender typicality. In this study, we examine whether ratings of the gender typicality of children's speech are affected by ratings of another potentially salient attribute that speech acoustics convey: speaker age. We conducted an experiment in which perceived gender and perceived age ratings were collected for the single-word productions of 57 5–13 year old children examined by Beaulieu *et al*. (18) and brief spoken narratives by the same children. Preliminary results suggest that the effects of talker sex are robust when both actual age and perceived age are controlled statistically.

3aSC23. Perception of prosody: How toddlers harness intonation to guide early attention. Jill C. Thorson (Commun. Sci. and Disord., Univ. of New Hampshire, 4 Library Way, Hewitt Hall, Dover, NH 03824, jill.thorson@unh.edu)

Young infants are born with language-specific preferences, particularly with respect to prosody (i.e., melody and rhythm). Previous work has shown that by 18-months, toddlers are guided by intonation and information status during an on-line reference resolution task (Thorson and Morgan, 14). This

study isolated the role of fundamental frequency (f0) during early attentional processing, showing that a bitonal f0 movement increases looking time to a target over a monotonal movement (and both show increased looking versus no pitch movements). The motivation for the current study is to examine the ability to perceive and utilize specific intonational patterns at earlier stages in speech and language development. The study asks whether typically developing 14-month-old toddlers are able to employ different intonational contours in order to attend to an object with unique information statuses (e.g., *new*, *given*). Methods include monitoring eye movements in response to varying pitch patterns and analyzing variables such as *total fixation time to a target* and *time of first fixation*. We hypothesize that at this early stage, toddlers will exploit the prosodic system to fixate on the discourse salient target. Critically, this work is a precursor to analyzing the early perception of prosody in young children with autism spectrum disorders.

3aSC24. Analysis of tongue trajectory variation for /r/ in older children and adults. Sarah Dugan (Psych., Univ. of Cincinnati, 67 Grady Court, Dayton, OH 45409, hamilsm@ucmail.uc.edu), Colin Annand (Psych., Univ. of Cincinnati, Cincinnati, OH), Sarah R. Li, Hannah M. Woeste, Jack A. Masterson (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Michael A. Riley (Psych., Univ. of Cincinnati, Cincinnati, OH), T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Kinematic studies of tongue articulator movement have found that older children and adolescents have smaller articulator displacements and greater movement variability compared to adults. However, many of these kinematic studies have used simple, early-developing phonemes as stimuli for comparison across talker groups. The American English rhotic /r/ is the latest to be acquired in children, possibly due to the complexity of the pharyngeal and oral constriction pattern required to produce the sound. In this study, we use automated processing of midsagittal ultrasound images to track trajectories of tongue blade, dorsum, and root during /r/ production in older children and adults. Our preliminary results suggest that the trajectories of tongue parts are different for adults and children, with children producing tongue part movements with greater relative amplitude and variability. We explore the possible developmental reasons for this outcome through the lens of dynamic principles of stability and change.

3aSC25. Secondary acoustic cues in adult perception of young children's stop productions. Elaine R. Hitchcock (Commun. Sci. and Disord., Montclair State Univ., 116 West End Ave., Pompton Plains, NJ 07444, hitchcocke@mail.montclair.edu) and Laura L. Koenig (Haskins Labs., New Haven, CT)

The perception of plosive voicing distinctions is attributed primarily to voice onset time (VOT). However, perceptual judgments of naturally produced stop tokens may be influenced by secondary cues, such as fundamental frequency (f0) and burst amplitude. The present study assessed the role of secondary cues in adult labeling judgments of bilabial and alveolar CV words produced by 2–3-year-old English-speaking children. Child productions were divided into three groups across the VOT continuum. Four exemplars per phoneme category (/p, b, t, d/) in each range (short, ambiguous, and long) were chosen from 6 children. Two stimulus sets were created. One set included pre-voiced tokens for /b d/ and exaggerated long lag tokens (100+ ms) for /p t/; the other set removed these extreme tokens. Listening data were obtained from adults per set. Listeners were highly accurate (>90%) at labeling the production as intended by the child when the production matched the VOT category expected for the target word. When VOT was ambiguous, judgments for voiced targets ranged from 75% to 92% and voiceless targets ranged from 54% to 62% accuracy. Measures of f0 and burst amplitude suggest that secondary cues in the speech signal contributed to adults' perception of children's stops.

3aSC26. Classification of accurate and error tongue movements for /r/ in children using trajectories from ultrasound. Sarah R. Li (Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, CVC, 3960, Cincinnati, OH 45241, lisr@mail.uc.edu), Sarah Dugan (Psych., Univ. of Cincinnati, Dayton, OH), Colin Annand (Psych., Univ. of Cincinnati, Cincinnati, OH), Hannah M. Woeste, Jack A. Masterson (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), and Michael A. Riley (Psych., Univ. of Cincinnati, Cincinnati, OH)

American English /r/ is considered the most difficult sound to remediate in children with speech sound disorders. The sound is complex, requiring coordinated, quasi-independent tongue movements, making it challenging to remediate in speech therapy. However, there are few clinical tools that allow for a quantitative, real-time display of the dynamics of tongue part movements. We aim to fill this need by developing a real-time, simplified, interactive biofeedback system that transforms ultrasound image data to a visual feedback object and quantitative data stream. In this presentation, we compare blade, dorsum, and root displacement trajectories determined from midsagittal ultrasound imaging of children with residual speech sound disorders (RSSD) and children with typical speech, using principal component models and cluster analysis techniques. Results show a trend of smaller tongue part displacements for RSSD children compared to children with typical speech. The analysis also elucidates distinct strategies for production of accurate /r/. We compare our classification of accurate and error tongue movements to perceptual judgments from trained listeners. Preliminary results suggest strong correspondence between our trajectory-based classification and listener judgments of /r/ accuracy.

3aSC27. Photomicrography of the middle ear ossicles. Matthew Kist and Peter M. Scheifele (Audiol., Univ. of Cincinnati, 3202 Eden Ave., Cincinnati, OH 45267, kistmj@mail.uc.edu)

I removed temporal bone sections from human cadavers with the help of my colleagues. From there, I utilized a Dremel tool to access the otic capsule and remove the malleus, incus, and stapes from each section. Not all attempts were successful due to the fragile nature of these bones. I extracted a total of 5 stapes, an incus, and 2 mallei in total. These bones were photographed using photomicrography: a 35 mm film camera attached to a microscope. The purpose of this research is to provide high quality images of the middle ear ossicles for educational purposes and learning tools. The ossicles will also be present in addition to the poster for individuals to hold and accurately grasp the size of these bones.

Session 3aSP

Signal Processing in Acoustics, Underwater Acoustics, Acoustical Oceanography, and Noise: Bayesian Inference in Acoustic Signal Processing

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

Zoi-Heleni Michalopoulou, Cochair
Mathematical Sciences, New Jersey Institute of Technology, 323 ML King Blvd, Newark, NJ 07102

Paul J. Gendron, Cochair
ECE Department, University of Massachusetts Dartmouth, 285 Old Westport Road, North Dartmouth, MA 02747

Chair's Introduction—8:30

Invited Papers

8:35

3aSP1. Physics-based distributions for randomly scattered signals and their Bayesian conjugate priors. D. K. Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr.-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D. Keith.Wilson@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), Vladimir E. Ostashev, Daniel J. Breton, and Carl R. Hart (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

Sound waves propagating through the atmosphere and ocean are randomly scattered by turbulence, internal waves, surface roughness, and other processes. In some limiting cases, probability density functions (pdfs) for the scattered signal variations can be derived, e.g., the log-normal pdf for weak scattering (in the Rytov approximation) and the exponential pdf for strong scattering. A variety of more general, usually empirically based, distributions are available which reduce to these limiting cases, such as the Rice, gamma, and generalized gamma. For situations involving multiple receivers, multivariate log-normal, Wishart, and matrix gamma pdfs may be employed. Parametric uncertainties and spatial/temporal variability in the scattering process can be addressed with a compound pdf formulation, which involves an additional distribution for the uncertain or variable parameters. From a Bayesian perspective, the scattering pdf corresponds to the likelihood function, the pdf for the uncertain parameters to the prior/posterior, and the compound pdf to the marginal likelihood. Many common scattering pdfs possess Bayesian conjugate priors, which lend themselves to simple updating equations and analytical solutions for the posteriors and marginal likelihoods. This presentation summarizes important pdfs for randomly scattered signals and their conjugate priors when available.

8:55

3aSP2. Bayesian inference when the physics is not quite right. Allan D. Pierce (Cape Cod Inst. for Sci. and Eng., P.O. Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann, and Elisabeth M. Brown (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Bayesian inference requires a physical and structural model for the portion of the environment being assessed. The model is characterized by a finite number of parameters, and these are assumed to each be a random variable. The forward problem can be solved given a knowldege of these parameters and the governing equations. Bayesian inference begins (prior probability) with some initial broad assumptions about the probability distributions. An intricate (Bayesian) process involving data and solutions to the forward problem leads to a refinement of these probability distributions. The range of possibilities becomes narrower, and one identifies most probable values of the parameters. The present paper raises the question as to what results when the parameterization is capriciously in conflict with what is known about the actual environment. An example from underwater acoustics is the choice of parameters to describe the frequency dependence of attenuation on a sediment layer. What is sometimes done is to assume the attenuation is directly proportional to frequency (just one parameter) or that the attenuation obeys a power law with a constant exponent (two parameters). The intrinsic shear modulus of the sediment is ignored in the numerical solutions of the forward problem, so the physics is incomplete. Some idealized examples are used to explore the consequences of such simplifying assumptions. Improved parameterizations are suggested that will yield more realistic frequency dependences of attenuation.

9:15

3aSP3. Trans-dimensional geoacoustic inversion and uncertainty quantification for SBCEX17 data. Stan E. Dosso (School of Earth & Ocean Sci., Univ. of Victoria, P.O. Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), Jan Dettmer (Univ. of Calgary, Calgary, AB, Canada), Charles W. Holland (Appl. Phys. Lab., The Penn State Univ., State College, PA), Josee Belcourt (School of Earth & Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Julien Bonnel (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Dag Tollefsen (Defence Systems Div., Norwegian Defence Res. Establishment, Horten, Norway), and David P. Knobles (Knobles Sci. and Anal., Austin, TX)

This paper presents an efficient and general approach to Bayesian inversion and uncertainty quantification for seabed geoacoustic profile estimation. The model-selection problem of estimating an appropriate seabed parameterization is addressed with trans-dimensional (trans-D) inversion via reversible-jump Markov-chain Monte Carlo, which samples probabilistically over the number of layers. An efficient proposal density for parameter perturbations is based on using a linearized approximation to the posterior probability density, applied in principal-component (PC) space where the (rotated) parameters are uncorrelated. The acceptance rate of perturbations and birth/death steps is improved by parallel tempering, based on a series of interacting Markov chains with successively tempered (relaxed) likelihoods. The PC proposals are adapted individually to the tempering of each Markov chain. The data-error model is based on the assumption of multivariate Gaussian errors with correlations represented by an autoregressive process. The parameters of zeroth- and first-order autoregressive error processes are sampled trans-dimensionally to avoid over- or under-parameterizing the error model. The approach is illustrated for three data sets from the 2017 Seabed Characterization Experiment (SBCEX17), including broadband seabed reflection coefficients; dispersion of water-borne acoustic modes, resolved by warping analysis; and ship noise recorded at a bottom-mounted horizontal array of hydrophones.

9:35

3aSP4. Employing geometric pattern discovery techniques for sonar target recognition. Ananya Sen Gupta (Elec. and Comput. Eng., Ctr. for the Eng. Arts and Sci., Univ. of Iowa, 4016 Seamans Iowa City, IA 52242, ananya-sengupta@uiowa.edu) and Ivars P. Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

Automated sonar target recognition suffers from linear superposition of acoustic color features, which need to be precisely disambiguated to correctly identify a target of the specific composition and geometry. Most target acoustic color images exhibit feature overlap between environmental reverberation as well as reflections due to target geometry, and elastic waves that are representative of the material composition of the target. From a signal processing perspective, this is essentially mitigating and disentangling interference from two highly competing yet hard-to-predict sources: reflections from the environment and reflections due to the target geometry, which represent different physical phenomena with distinct geometric patterns. We will present recent results in feature selection and interference mitigation employing Gabor wavelets to separate target acoustic color features due to elastic wave orbits from interfering features due to the target geometry and environmental effects. We will provide results from simulations as well as experimental field data and discuss the trade-offs in feature representations employing acoustic color and those that also include the phase information. Finally, we will discuss how this type of feature extraction can be used to discover and create robust priors for Bayesian inference networks for learning salient target features.

9:55

3aSP5. Hybrid Cramer-Rao bound for near-field source localization using a spatially spread acoustic vector sensor. Zakayo Ndiku Morris (Dept. of Electron. and Information Eng., The Hong Kong Polytechnic Univ., Hung Hom KLN, Hong Kong, zakayo.morris@connect.polyu.hk), Kainam Thomas Wong (none, San Gabriel, CA), and Yuting Han (Data Sci. Program, Worcester Polytechnic Inst., Worcester, MA)

This presentation characterizes the source-localization precision obtainable via received signal strength indication (RSSI) based on data from a tri-axial velocity sensor and a spatially separated pressure sensor. That scheme was proposed originally by Y. I. Wu and K. T. Wong in January 2012 in the *IEEE Transactions on Aerospace and Electronic Systems*. That source-localization scheme depends on the acoustic propagation path-loss exponent, which is typically not precisely known a priori but could be modeled stochastically. That exponent may also differ in value for the propagation path to the tri-axial velocity sensor and for the propagation path to the pressure sensor. This presentation accounts for these two practical considerations in characterizing the scheme's source-localization precision, through the metric of the hybrid Cramer-Rao bound (HCRB), the correctness of which is here validated by Monte Carlo simulations of the corresponding maximum a posteriori (MAP) estimator.

10:15–10:30 Break

10:30

3aSP6. Bayesian inference for water column and sediment inversion in ocean acoustics. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

Bayesian methods provide inference on unknown parameters given prior beliefs and a statistical model for the observations in a noisy environment. It was shown in "Sequential filtering and linearization for inversion in the Seabed Characterization Experiment," *J. Acoust. Soc. Am.* **144**, 1913 (2018) how geometry and sediment parameter estimation in an oceanic experiment can be achieved with a two-stage Bayesian approach. Here, we demonstrate the potential of the method for inversion with a different data set presenting challenges in identifying features of the acoustic field necessary for the second stage of the estimation process. The water column sound speed is complex in this experiment, necessitating a complex inverse model. The transmitted signals are lfm pulses and probability densities of multipath arrival times are calculated at the receivers. For the forward and inverse calculations, we compute the Jacobian matrix for sound speed estimation using EOF coefficients. The inversion, employing the direct and surface and bottom paths, leads to inference on the geometry and the water column sound speed profile. Using the computed probability densities and introducing one more arrival, sediment property densities are calculated. Results are in excellent agreement with expected values for the site. [Work supported by ONR.]

10:50

3aSP7. Localization of a mobile acoustic scatterer from sub-Rayleigh resolved acoustic arrivals in a refractive environment. Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, North Dartmouth, MA) and Abner C. Barros (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Darmouth, MA 02747, abarros1@umassd.edu)

Computational Bayesian inference on the state of an underwater mobile object from a continuous active acoustic transmission is presented. The challenge of sub-Rayleigh resolvable wave vectors in refractive environments with uncertainty in ambient acoustic noise power is addressed. The location and speed of the mobile scatterer are inferred under the challenging constraint of a small receive vertical aperture. The need for jointly inferring the vertical angles and Doppler offsets of the arrivals is addressed with a Gibbs sampling approach. The posterior density of the plane wave components is mapped to the object's range, depth, and speed through ray interpolation. A case scenario from an acoustic duct environment in the western Indian ocean is presented.

11:10

3aSP8. Bayesian framework for direction of arrival analysis using spherical harmonics. Stephen Weikel (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY 12180, weikes@rpi.edu), Christopher Landschoot (Kirkegaard Assoc., Chicago, IL), and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

A common task in acoustical applications is the determination of directions of arrival (DoAs) of sound at a receiver. This work aims to address this problem in situations involving potentially multiple simultaneous sound sources by means of a two-level framework of Bayesian inference. This process involves first estimating the number of sound sources present, followed by estimating their directional information, based on sound data collected with a spherical microphone array. Analytical models are formulated using spherical harmonic beamforming techniques, which are incorporated into the Bayesian analysis as part of the prior information. The experimental data are also incorporated to update the information available prior to analysis. All necessary prior information is assigned based on the principle of maximum entropy. Through this technique, the number of sources is first estimated, and then, the DoA information of those sources is extracted from the most concise model that adequately fits the experimental data. This paper presents the Bayesian formulation and analysis results to demonstrate the potential usefulness of model-based Bayesian inference for determining DoAs in complex noise environments with potentially multiple concurrent sources.

Contributed Paper

11:30

3aSP9. Sparse Bayesian learning beamforming combined with short-time Fourier transform for fault detection of wind turbine blades. Xiang Pan, Zefeng Cheng, Zheng Zheng, and Yehui Zhang (School of Information and Electron. Eng., Zhejiang Univ., Zhe Rd. 38, Hangzhou, Zhejiang 310027, China, panxiang@zju.edu.cn)

Wind energy is one of the important renewable energy resources. The wind turbines need to be checked every now and then to enhance security. The rotor blade of the wind turbine can be damaged due to long-term running in harsh environments and complex vibrations resulting in the crack of blade. For detection of the blade faults, the sparse Bayesian learning (SBL) beamforming (BF) is implemented to the acoustic data received by a microphone array on the ground for signal enhancement. The direction of arrival of the abnormal sound can be estimated with high resolution meanwhile interferences, such as noise emitted by cooling fans, can be suppressed by the low sidelobes provided by the SBL-BF. After the Short-Time Fourier Transform (STFT) is carried out over the enhanced signals, it is seen from the time-frequency spectrum that the abnormal sound appears with an approximate 6-s cycle. By detecting the cyclical characteristics, one can decide whether there is or not the blade fault. A real-time processing system for detection of the blade faults underwent a number of tests in a coastal plain, a hilly area, and the Xinjiang Plateau. The test results have demonstrated the effectiveness of the blade fault detection framework.

Session 3aUW

Underwater Acoustics and Acoustical Oceanography: Ocean Acoustics in High Latitudes and General Propagation

Mohsen Badiey, Chair
University of Delaware, University of Delaware, Newark, DE 19716

Contributed Papers

9:00

3aUW1. Azimuthal dependence of the acoustic field in a year long Canada Basin Acoustic Propagation Experiment. Mohsen Badiey (Univ. of Delaware, University of Delaware, Newark, DE 19716, badiey@udel.edu), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Sean Pecknold (Defence Res. and Development Canada, Dartmouth, NS, Canada), Megan S. Ballard, Jason D. Sagers (Appl. Res. Lab.Univ. of Texas, Austin, TX), Altan Turgut (Naval Res. Lab., Washington, DC), John A. Colosi (Oceanogr., Naval Post Graduate School, Monterey, CA), Peter F. Worcester, and Mathew Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

As in the mid-latitudes, the variability of the oceanography in the arctic ocean causes azimuthal dependent acoustic fields induced by the environmental parameters, such as the variable sound speed profile and bathymetry. Preliminary analysis of a year-long experimental data from "deep-to-shallow" and the "shallow-water" Canada Basin Acoustic Propagation Experiment (SW CANAPE) shows strong azimuthal variability of broadband signals recorded on 11 spatially distributed acoustic receiver arrays on the Chukchi shelf from September 2016 to October 2017. Particular attention is paid on geotimes pertaining to the variable sound speed profiles around the dynamic shelf-break region. Although acoustic modeling of the propagation requires a physics based ocean model for a complete modeling exercise, preliminary determination of the azimuthal behavior of broadband propagation is conducted using multiple two-dimensional Parabolic Equation (Nx2D) with the collected oceanographic data. Calculations for long-range, as well as short-range, source-receiver pairs in the SW CANAPE data show that azimuthal variabilities occur frequently and are related to the spatial and temporal characteristics of signal dispersion for different range and geotime scales. [Work supported by ONR.]

9:15

3aUW2. Temporal and spatial dependence of a yearlong record of sound propagation from the Canada Basin to the Chukchi Shelf. Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Mohsen Badiey (Univ. of Delaware, Newark, DE), John A. Colosi (Naval Postgrad. School, Monterey, CA), Altan Turgut (U.S. Naval Res. Lab., Washington, DC), Sean Pecknold (Defence Res. and Development Canada, Dartmouth, NS, Canada), Ying-Tsong Lin, Andrey Y. Proshutinsky, Richard A. Krishfield (Woods Hole Oceanographic Inst., Woods Hole, MA), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

During the Canada Basin Acoustic Propagation Experiment (CANAPE), low-frequency signals from five tomographic sources located in the Canada Basin were recorded by an array of hydrophones located on the Chukchi Shelf. The propagation distances ranged from 240 km to 520 km, and the propagation conditions changed from persistently ducted in the basin to seasonally upward refracting on the continental shelf. An analysis of the received level from the tomography sources revealed a spatial dependence in the onset of the seasonal increase in transmission loss, which was correlated with the locations of the sources in the basin. This observation led to the hypothesis that the water advected from Barrow Canyon westward over the continental slope by the Chukchi slope current contributes to the temporal and spatial dependence observed in the acoustic record. The water column properties and ice draft were measured by oceanographic sensors on the basin tomography moorings and by six arrays of oceanographic moorings on the continental shelf to characterize the temporal and spatial variability of the environment. This talk examines the range-dependent measurements and explains the observed variability in the received signals through propagation modeling. [Work sponsored by ONR.]

9:30

3aUW3. Seabed properties on the Chukchi Shelf observed during the 2016–-2017 canada basin acoustic propagation experiment. Jason D. Sagers, Megan S. Ballard (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu), and Sean Pecknold (Defence Res. and Development Canada, Dartmouth, NS, Canada)

Previous work examined seabed geoacoustical properties near the 150 m isobath on the Chukchi shelf as observed during the 2016–2017 Canada Basin Acoustic Propagation Experiment (CANAPE) [Sagers and Ballard, *JASA* **144**(3), 1666]. That work reported a water/sediment compressional sound speed ratio near 0.98 and a compressional sound speed gradient in the upper sediment around 6 s^{-1}. This work extends the prior analysis by investigating additional locations throughout the larger shallow-water experimental site. Data from subbottom profile surveys, measurements from an *in situ* acoustic coring system and inferences made using ship-radiated noise received on Autonomous Multichannel Acoustic Recorders (AMARs) deployed by the Defence Research and Development Canada (DRDC) are examined to understand seabed layering and sediment properties throughout the experimental region. Particular interest is given to understanding whether geoacoustical properties on the Chukchi shelf exhibit range-dependence. [Work sponsored by ONR.]

9:45

3aUW4. Modeling mid-frequency reverberation in the Arctic Ocean. Dajun Tang (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

With renewed interest in the Arctic in response to its changing environment, various acoustic field measurements are underway or being planned. Anticipating the utility of reverberation as a tool to probe the environment, as well as to detect targets, here a time-domain reverberation model suitable for the frequency band of 0.5—5 kHz is introduced. This model assumes that the sea ice as the scattering mechanism and sea ice scattering parameters are taken from a set of laboratory measurements. It is found that the double-duct sound speed profile in the Arctic enables sound to propagate to long distances, and the main loss mechanism is scattering by sea ice, including ice keels. Reverberation measured at different azimuths is hypothesized

as an effective method to estimate angular-dependent keel density and varying sound channels. [Work supported by ONR Ocean Acoustics.]

10:00

3aUW5. Forty-year review of developments in underwater acoustic modeling. Paul C. Etter (Northrop Grumman Corp., P.O. Box 1693, Baltimore, MD 21203, paul.etter@ngc.com)

This is the sixth paper in a series of reviews presented at eight-year intervals starting in 1979 [*J. Acoust. Soc. Am.* **65**, S42 (1979); **82**, S102 (1987); **97**, 3312 (1995); **114**, 2430 (2003); **129**, 2631 (2011)]. All surveys covered basic acoustic models and sonar performance models. Basic acoustic models included 143 propagation, 29 noise, and 32 reverberation models. Propagation models were categorized according to ray theory, multipath expansion, normal mode, fast field, and parabolic approximation formulations; further distinctions were made between range-independent and range-dependent solutions. Noise models included ambient noise and beam-noise statistics models. Reverberation models included cell and point-scattering formulations. The 44 sonar performance models included active sonar models, model operating systems, and tactical decision aids. Active sonar models integrated basic acoustic models, signal-processing models, and supporting databases into cohesive operating systems organized to solve the sonar equations. Model operating systems provided a framework for the direct linkage of data-management software with computer-implemented codes of acoustic models. Tactical decision aids represented a form of engagement-level simulation that blended environmental and sonar performance information with tactical rules. The overall inventory increased by 5 models per year; historical references are maintained in the fifth edition of *Underwater Acoustic Modeling and Simulation*.

10:15–10:30 Break

10:30

3aUW6. Shallow water acoustic propagation prediction. Cathy Ann Clark (Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell St., B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

A set of propagation loss curves, extracted from measured reverberation in an environment with water depths of 200–250 ft is used to investigate the crossover region between deep and shallow water using two normal mode implementations designed to predict propagation in deep and shallow water, respectively. A third range-recursive calculation which is applicable in situations for which cycle mixing with range results in randomization of phase interference between modes is also used for comparison. The impact of forward boundary scattering on reproducing the measured levels at multiple receiver depths is demonstrated.

10:45

3aUW7. Modeled position and intensity statistics in deep water caustics. Katherine F. Woolfe (Leidos, 672 Brookline St. SW, Atlanta, Georgia 30310, katherine.woolfe@gmail.com), Chuck Spofford (Leidos, Fairfax, VA), Kritika Vayur (Leidos, State College, PA), and Peter Mikhalvesky (Leidos, Arlington, VA)

Caustics are a form of natural focusing that occurs in deep water acoustic propagation, but acoustic intensity and position perturbations at caustics have not been investigated in detail. The positions, shapes, and intensities of caustics and cusps are controlled primarily by the large-scale sound speed profile, with perturbations primarily caused by deterministic features (fronts, eddies) and stochastic features (internal waves). We selected two representative deep water environments for our analysis: a north Pacific site and temperate profile. For each region, 100 independent and 100 time-coherent realizations of internal waves were generated. The model uses the statistics provided by the Garrett-Munk power spectrum to generate displacements, which are converted into sound speed realizations. In the north Pacific profile, internal waves were concentrated primarily in the upper 200 m of water. In the temperate profile, internal waves were concentrated in the upper 700 m of water. We used a Gaussian beam model to analyze position and intensity statistics of caustics for both types (independent and time-coherent) of internal wave fields at each site as a function of frequency, internal wave strength, and source depth. Simulations indicate that caustics are more stable than other portions of the acoustic field and that this stability is a function of internal wave strength and caustic location. This work is approved for public release, Distribution Unlimited.

11:00

3aUW8. Analytic solution to a waveguide featuring caustics and shadow zones. Brian M. Worthmann (Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94550, bworthma@umich.edu)

Exact solutions to waveguides are useful tools for validating numerical waveguide models. In this paper, a modal solution to an unbounded waveguide with a range-independent but depth-dependent profile is described. The sound speed profile utilized is a symmetric, piecewise n^2-linear profile matched above and below to a homogenous half-layer with density and sound speeds that prevent any reflections at the interface. In this environment, caustics and shadow zones are formed. A nearly exact solution to this environment could help improve other numerical methods' accuracy near caustics and shadow zones. In this mathematical analysis, both proper (trapped) and improper (leaky) modes are included. Arbitrarily high frequencies are permitted through the use of numerically well-conditioned approximations to Airy functions, particularly for the dispersion relations and mode-shape evaluations. The proper (real) and improper (complex) modal eigenvalues are found approximately, with root-finding algorithms used for eigenvalues near cut-off to improve the approximation. Mathematical details are presented along with a brief discussion of how well a few common numerical solvers agree with the analytic solution derived.

11:15

3aUW9. Preliminary soundscape observations from a Colombian humpback whale breeding ground before port construction. Kerri Seger (Appl. Ocean Sci., 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Christina E. Perazio (Univ. at Buffalo, Biddeford, Maine), Valeria Gonzalez (Pacifico Adventures, Buenos Aires, Argentina), Andrea Luna-Acosta (Pontificia Universidad Javeriana, Bogota, Colombia), and Natalia Botero (Fundación Macuáticos Colombia, Medellin, Colombia)

The Gulf of Tribugá in the Colombian Pacific is still relatively undisturbed. No road access between villages nor from major urban areas exists. Small boats and walking beaches at low tide serve as the main transportation conduits. In this breeding ground for humpback whale Stock G, small-scale artisanal and shrimp fisheries and whale-watching activities support the livelihood of local communities but may already interfere with biological communication systems. Once a proposed international port is built anthropogenic pressure is expected to increase. Baseline information is key to understand the Gulf's current acoustic state and to document the best approximation of its practically pristine state. An ecological Acoustic Recorder (EAR, Oceanwide Science Institute) was deployed between the largest town (about 2000 residents) and the southern boundary of the Utria National Park marine reserve from October to December, 2018. Also, opportunistic over-the-side recordings from July to September, 2018, provided point surveys at a greater spatial resolution during peak humpback whale breeding season. Results include the first acoustic catalogue of the area and a preliminary understanding of important bandwidths for future monitoring and current contributions of small vessels to the soundscape.

11:30

3aUW10. Characterization of sound channel axis and depth in the global ocean. Mukunda Acharya and Likun Zhang (National Ctr. for Physical Acoust., Univ. of Mississippi, 112-, 114 Chucky Mullins Dr., Oxford, MS 38655 , mkachary@go.olemiss.edu)

Sound speed in the ocean has a minimum at a certain depth that acts as the axis of sound channels for underwater sound propagation. In this work, we characterize the three dimensional features of the axial sound speed and depth in the global ocean. The characterization follows from calculation of more than 16 000 sound speed profiles by using the conductivity, temperature, and pressure data in the World Ocean Circulation Experiment. The axial speed and depth as a function of latitude are determined for the Atlantic, Pacific, and Indian oceans, and a polynomial fitting for the dependence is provided. The variability of axial sound channel and depth between

different oceans and latitude is identified and related to the feature of the thermoclines of each ocean. The characterization and formulation provide knowledge of the sound channel in the global ocean that could be particularly useful for modeling long-range and three-dimensional underwater sound propagation for applications in underwater acoustic tomography and monitoring seismic activities.

WEDNESDAY AFTERNOON, 15 MAY 2019 FRENCH, 1:00 P.M. TO 2:35 P.M.

Session 3pAA

Architectural Acoustics: Acoustical Materials and Testing

Benjamin Bridgewater, Cochair
Architecture, University of Kansas, 1536 Ogden Street, Denver, CO 80218

Shane J. Kanter, Cochair
Threshold Acoustics, 53 W. Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—1:00

Contributed Papers

1:05

3pAA1. Broadband design of multilayer micro-slit panel absorbers for improved transparency using Bayesian inference. Michael Hoeft, Cameron J. Fackler, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, hoeftm@rpi.edu)

Multilayer Micro-Slit panels (MSP) are assessed for their potential as broadband absorbers that simultaneously maintain visual transparency. The late Dah-You Maa introduced micro-Slit panel absorbers as a continuation of his previously developed Micro-Perforated panel (MPP) absorbers. Like MPPs, Micro-Slit panels allow high absorption coefficients to be achieved without using fibrous materials but are limited to a narrow frequency bandwidth. Broadband absorption can be achieved by combining panels into a multilayer absorber. The complexity of determining the optimum set of parameters required to fulfill a design scheme necessitates implementation of the Bayesian framework. This probabilistic method automatically determines the most concise number of layers required and gives parameters for each layer of the resulting composite. Various slit patterns and algorithms introduce a family of solutions, which satisfy the design scheme, for optimization of visual transparency. The Micro-Slit samples can be fabricated and tested to validate acoustic performance and assess visual properties.

1:20

3pAA2. Studies on sound absorption performance of porous materials at low temperature. Xiwei Wang (HuaXinSiFang (Beijing) Construction Technol. Co., Ltd., Beijing, China), Xiang Yan (Acoust. Lab of School of Architecture, Tsinghua Univ., Beijing, China), Hui Li (Beijing Deshang Jingjie Technol. Development Co., Ltd., Main Bldg., Rm. 104, Beijing 100084, China, lihuisylvia@aliyun.com), and Xiaoyan Xue (HuaXinSiFang (Beijing) Construction Technol. Co., Ltd., Beijing, China)

The acoustical parameters of porous materials are usually tested at normal temperature. In recent years, spaces need acoustical treatments at low temperature spring up with the dynamic tendency of projects such as snow theme parks and high speed trains traveling in the frozen region. The sound absorption coefficient of porous materials at low temperature is very rare for now. So in real projects, the sound absorption coefficient of porous materials at normal temperature is the only reference which is certainly inappropriate. The sound absorption coefficient of several common porous materials is test at $-50°C$, $-25°C$, and $25°C$. The details of the measurement process and data analysis are specified in this paper.

1:35

3pAA3. The spectrum of concrete: Material research and experimentation expanding previous sound lab results for various acoustical environments. Daniel Butko (Architecture, The Univ. of Oklahoma, 830 Van Vleet Oval, Norman, OK 73019, butko@ou.edu)

This paper presentation continues progress of an active multiphase acoustical research project emphasizing implementation of prototypical concrete systems in multiuse spaces to transform unhealthy learning environments. Building upon research results shared during the December 2017 and November 2018 ASA paper presentations, various admixtures/additives are being explored as viable ingredients to develop concrete mixtures capable of decreasing excessive reverberation and flutter echo while increasing clarity and speech intelligibility. Faculty and students, working in both physical and digital realms with industry partners, are analyzing and validating consequences of manipulating materials for resultant acoustical criteria. The work presented here showcases research and development of how concrete can be processed, shaped, surfaced, and implemented into full-scale built form to provide scholarly merit applicable in/with similar spaces and/or functions.

1:50

3pAA4. Impact sound isolation measurement analysis for 6 typical type of floors in China. Xiaoyan Xue (HuaXinSiFang (Beijing) Construction Technol. Co., Ltd., Beijing, China), Hui Li (Beijing Deshang Jingjie Technol. Development Co., Ltd., Main Bldg., Rm. 104, Beijing 100084, China, lihuisylvia@aliyun.com), and Xiang Yan (Acoust. Lab of School of Architecture, Tsinghua Univ., Beijing, China)

Single layer homogeneous elastic material floating floor, porous vibration isolation material floating floor, vibration isolation cube floating floor, spring floating floor, floor with hard finish topping, and floor with soft finish

topping are 6 types of impact sound isolation floor commonly used in China. All the 6 types of floor are test in laboratory by both light impact method with standard tapping machine and heavy impact method with rubber tire and rubber ball. Data analysis is focused on one hand the difference between test methods for the same floor and the other hand the sound isolation performance of different floors by the same test methods.

2:05

3pAA5. A high-resolution goniometer to measure sound diffraction patterns. Anthony Savino, Jonathan Kawasaki, and Ning Xiang (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, savina@rpi.edu)

Surface scattering and diffuse reflections from acoustic diffusers have recently become a significant topic of research for room acoustics. An acoustical goniometer can be implemented to create a circular microphone array to characterize the polar response for various types of diffusers. Particular areas of interest include validating the Physical Theory of Diffraction (PTD) through the use of finite-sized diffusers. In order to properly describe this phenomenon, a microphone array must achieve a fine enough angular resolution in addition to fulfilling the far-field requirements. This talk will discuss the implementation and results of a portable goniometer with a radius of 5 m and an angular resolution of 1.25 deg. This design is intended to be easily deployed in empty, indoor spaces of sufficiently large dimensions for a full 360-deg diffraction response measurement.

2:20

3pAA6. Adaptive reuse of unconventional spaces for performance: Music in a museum and opera in a warehouse. Samuel H. Underwood (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE), Kieren H. Smith (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Brigham Young University, Provo, UT 84602, kierenhs@gmail.com), and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, Omaha, NE)

Taking a space designed for one purpose and creating an acoustically sound space for musical performance presents unique challenges. A local opera company has undertaken two projects that involve such modifications for this performance season and have sought guidance on the unique acoustic complications involved. The first project was a percussive musical playground in a large art museum, adjacent to sound-sensitive exhibits. The second involved the adaptation of an existing warehouse into a space suitable for the performance of a ballet-opera. Researchers worked with set designers early in the process and throughout set construction. In order to better understand the spaces, impulse responses were taken and analyzed. Impulse response measurement results, acoustic models with suggested modifications, and auralizations at key listening positions are presented to address unique challenges associated with these unconventional spaces. An emphasis has been placed on reaching target values for reverberation time within limited time and resource constraints. Additionally, sound isolation for the two spaces has been addressed with special regard to adjacencies and noises in industrial downtown locations.

WEDNESDAY AFTERNOON, 15 MAY 2019 CLEMENTS, 1:30 P.M. TO 2:45 P.M.

Session 3pAB

Animal Bioacoustics: Topics in Animal Bioacoustics

Christopher Biggs, Chair
Marine Science Institute, University of Texas, 750 Channelview Dr., Port Aransas, TX 78373

Contributed Papers

1:30

3pAB1. Bald eagles (*Haliaeetus leucocephalus*) monitor their immediate acoustic environment vigilantly. JoAnn McGee, Peggy B. Nelson, Jeff Marr, Julia Ponder, Christopher Milliren, Christopher Feist, Andrew Byrne, and Edward J. Walsh (Univ. of Minnesota, S39 Elliott Hall, 75 East River Parkway, Minneapolis, MN 55455, mcgeej@umn.edu)

As part of an initial effort to determine if acoustic signals can be used to discourage eagles from entering wind turbine facility airspaces and thereby reduce morbidity/mortality collision rates, behavioral responses of bald eagles (*Haliaeetus leucocephalus*) to a battery of both natural and synthetic acoustic stimuli of varying spectral complexities were studied. Each signal was directed randomly to one of two loudspeakers in a sequence of ~10 trials, and the stimulus order was randomized for each bird. A variant of an observer-based psychoacoustic protocol was implemented, and judges were instructed to report the absence or presence of a response, response strength, and other distinctive response attributes. In pilot studies, subjects responded to ~74% of trials across all stimuli. Responsivity was greater to spectrally complex stimuli (~80% vs 59%), and greater responsivity was observed to natural stimuli than to synthetic stimuli (~82% vs 69%). Responsive subjects oriented correctly in the direction of the signal source in ~74% of trials. A significant difference in overall responsivity was not observed across stimulus sets, although habituation was observed across repeated trials when responses to all stimulus types were combined. The relevance of findings in relation to the design of deterrence/alerting protocols will be discussed. [Work supported by DOE Grant No. DE-EE0007881.]

1:45

3pAB2. The effects of age and sex on rates of hearing loss for pure tones varying in duration. Anastasiya Kobrina and Micheal L. Dent (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu)

Mice are frequently used to study and model presbycusis due to similarities in the human and mouse cochleae and in genetic makeup. Most of the previous research on presbycusis used electrophysiological measures of hearing in mice, leading to an underrepresentation of behavioral experiments in the literature on mouse aging. The goal of the current research was

to fill this gap by behaviorally measuring audiograms and temporal summation functions in aging mice. Adult mice were trained and tested using an accelerated longitudinal design. Mice were trained on a detection task using operant conditioning procedures with positive reinforcement. Audiograms were constructed using thresholds for 8, 16, 24, 42, and 64 kHz pure tones. Temporal summation functions were constructed for 16 and 64 kHz pure tones ranging from 20 to 800 ms in duration. The results revealed that mice retain pure tone hearing late into their lifespan, with high-frequency hearing loss preceding low-frequency hearing loss. Mice also benefit from increases in the duration of pure tones; however. this benefit decreases with age. Generally, male mice lose hearing at a faster rate than females. These results highlight the importance of measuring hearing in awake, trained, behaving subjects when comparing presbycusis across species.

2:00

3pAB3. Arabian horse vocalization: A brief look at romance. David Browning (Browning Biotech, 139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com), Peter Herstein (Browning Biotech, Westerly, RI), and Peter M. Scheifele (Univ. of Cincinnati, Cincinnati, OH)

Arabian horses are an expressive breed. It has been determined that domestic horse whinnies have two unrelated fundamental frequencies, termed F(0) and G(0), with their harmonics such as F(2), etc. F(0) can be connected to arousal, while G(0) to mood. Fifteen Arabian whinnies during quiet barn conditions were analyzed to determine typical calm F(0) and G(0) values. A limited number of whinnies were than obtained from a corralled stallion during his recognition, anticipation, and culmination of a mare being brought to nuzzling distance on the other side of the fence. The spectra of his prenuzzling whinnies are as might be expected but after the culmination he gives a loud whinny with a complex structure that is not totally understood.

2:15

3pAB4. Propagation loss of spawning calls produced by spotted seatrout *Cynoscion nebulosus* and the effective detection area of passive acoustic sampling. Christopher Biggs (Marine Sci. Inst., Univ. of Texas, 750 Channelview Dr., Port Aransas, TX 78373, cbiggs@utexas.edu), Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, Austin, TX), and Brad Erisman (Marine Sci. Inst., Univ. of Texas, Port Aransas, TX)

Acoustic signaling in fish has been observed in conjunction with various behaviors but most commonly is associated with spawning. Sound production has been used to identify spawning sites for multiple species, but sound propagation characteristics are often not accounted for, severely limiting the spatial resolution at which spawning sites can be identified. We examined the propagation loss of seatrout calls in the very shallow environment of an estuary using recorded calls and an array of stationary hydrophones. We estimated the minimum expected source level of a seatrout call from the maximum recorded sound pressure level (SPL) of individual fish calls made *in situ* at spawning sites. Based on preliminary data from 86 samples, the minimum expected source level of an individual call is 142.1 $\pm$ 0.9 (CI_{95}) dB_{rms} *re*: 1 μPa at 1m. Measurements of propagation loss were fitted with a logarithmic model, and preliminary results showed that seatrout calls were detectable above background noise levels between 64 m and 512 m from the source. These results can be used to relate sound levels to the abundance of fish present at a spawning aggregation and enhance the precision of locating spawning sites using passive acoustics.

2:30

3pAB5. Sex-differences in timing of the black-capped chickadee fee-bee song. Anastasiya Kobrina (Psych., SUNY Univ. at Buffalo, B23 Park Hall, Amherst, NY 14261, akobrina@buffalo.edu), Allison H. Hahn (Psych., St. Norbert College, Edmonton, AB, Canada), Eduardo Mercado (Psych., SUNY Univ. at Buffalo, Buffalo, NY), and Christopher B. Sturdy (Psych., Univ. of Alberta, Edmonton, AB, Canada)

Male black-capped chickadees produce *fee-bee* songs in spring for mate attraction and territorial defense. Less is known about the female song use in this species although a version of song, soft-song, appears to be used in mate-mate communication. Recent analyses of songs produced by chickadees revealed that female *fee-bee* songs are distinct from male songs in the spectral domain. Chickadees also precisely control the timing of their *fee-bee* songs during territory defense. No previous work has explored whether there are sex differences in the temporal patterning of *fee-bee* song production. Inter-song intervals were extracted from recordings of *fee-bee* songs produced in non-social contexts by 8 male and 7 female birds. Male chickadees produced *fee-bee* songs regularly, with the majority of songs spaced at intervals of 2.5–5.0 s. Song timing by females was more variable with production intervals ranging from 1.5 to 8.0 s. The relative stereotypy of song timing by males is consistent with earlier work suggesting that males may modulate song timing to communicate with other birds (e.g., by timing song production to either reduce or increase the likelihood that their songs overlap with those of other singers), and with differential use of songs by males and females.

Session 3pBAa

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Competition (Poster Session)

Kevin J. Haworth, Chair
University of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45209

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with USD $500 for first prize, USD $300 for second prize, and USD $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee.

Below is a list of students competing, with abstract numbers titles. Full abstracts can be found in the oral sessions associated with the abstract numbers. All entries will be on display, and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

2aBA2. Live color encoded speckle imaging platform for real-time complex flow visualization *in vivo*
Student author: Billy Y. S. Yiu

2aBA3. Design of carotid bifurcation phantoms for integrative imaging investigations of arterial wall and flow dynamics
Student author: Adrian J. Y. Chee

2aBA4. Atherosclerosis characterization using lipid-specific photoacoustic imaging and 4D ultrasound strain mapping in mice
Student author: Gurneet S. Sangha

2aBA5. Spatial analysis of cardiac strain using high-frequency four-dimensional ultrasound in mice
Student author: Frederick William Damen

2aBA6. Quantification of murine cardiac hypertrophy using 4D ultrasound
Student author: Alycia G. Berman

2aBA10. Ascertaining the relationship between acoustic droplet vaporization, inertial cavitation, and hemolysis
Student author: Newsha Jahanpanah

2aBA11. Frequency dependence of the vaporization threshold of sonosensitive perfluorocarbon droplets varying their liquid core and size
Student author: Mitra Aliabouzar

2aBA12. Acoustic droplet vaporization with microfluidic droplets results in dissolved oxygen scavenging
Student author: Rachel P. Benton

2pBA4. Detection of nucleic acid-loaded microbubbles in mouse hearts during ultrasound-mediated delivery
Student author: Meghan R. Campbell

2pBA5. Pentagalloyl glucose effects on murine abdominal aortic aneurysms
Student author: Jennifer L. Anderson

3aBA4. Spectroscopic photoacoustic imaging for cardiovascular interventions
Student author: Sophinese Iskander-Rizk

3pBAb5. Evaluation of a potential medical diagnosis application of sonoluminescence
Student author: Alicia Casacchia

3pBAb6. Deep-learning framework and acoustic reflector for improved limited-view and sparse photoacoustic tomography
Student author: Irvane Ngnie Kamga

3pBAb7. Pixel-wise deep learning for improving image reconstruction in photoacoustic tomography
Student author: Steven Guan

4aBAa7. Super-resolution ultrasound imaging for in vivo microvasculature assessment in acute kidney injury mouse model
Student author: Qiyang Chen

4aBAa8. Efficient sub-diffraction passive cavitation imaging
Student author: Scott J. Schoen

4aBAa10. Echo-mode aberration tomography: Sound speed imaging with a single linear array
Student author: Anthony Podkowa

4aBAa13. Inferring elastic moduli of drops in acoustic fields
Student author: Jesse Batson

4aBAb1. A parametric evaluation of shear wave speeds estimated with the time-of-flight approach in viscoelastic media
Student author: Luke M. Wiseman

4aBAb2. Measured power law attenuation of shear waves in swine liver
Student author: Steven A. Grosz

4aBAb4. Approximate analytical time-domain Green's functions for space-fractional wave equations
Student author: Madison Carriere

4aBAb5. Validity of independent scattering approximation (ISA) to measure ultrasonic attenuation in porous structures with mono-disperse random pore distribution
Student author: Yasamin Karbalaeisadegh

4aBAb7. Investigation into tendon histotripsy
Student author: Molly Smallcomb

4aBAb9. Design of a histotripsy array for the treatment of intracerebral hemorrhage
Student author: Tyler Gerhardson

4aBAb11. *Ex vivo* thermal ablation monitoring using three-dimensional ultrasound echo decorrelation imaging
Student author: Elmira Ghahrahmani Z.

4pBAa1. Photoacoustic tomography in a clinical linear accelerator for quantitative radiation dosimetry
Student author: David A. Johnstone

4pBAa2. Comparisons of inverse and forward problem approaches to elastography
Student author: Siavash Ghavami

4pBAb5. Microstructural anisotropy evaluation in trabecular bone structure using the mode-converted (longitudinal to transverse, L-T) ultrasonic scattering
Student author: Omid Yousefian

4pBAb9. Standing acoustic waves in microfluidic channels for enhanced intracellular delivery of molecular compounds
Student author: Connor S. Centner

4pBAb12. Development and characterization of acoustically responsive exosomes for simultaneous imaging and drug delivery applications
Student author: Jenna Osborn

4pBAb14. Optimization of molecular delivery to red blood cells using an ultrasound-integrated microfluidic system
Student author: Emily Margaret Murphy

5aBA1. Electroacoustic tomography system using ultra-short electric filed excitation source induced acoustic signals
Student author: Ali Zarafshani

5aBA3. Tissue Doppler imaging to detect muscle fatigue
Student author: Joseph Majdi

5aBA7. Loudness growth as a function of presentation method: Comparison of normal hearing children with children using cochlear implants
Student author: Shubha Tak

5aBA9. Renal volume reconstruction using free-hand ultrasound scans
Student author: Alex Benjamin

5aBA10. Etiology of the color Doppler twinkling artifact on kidney stones
Student author: Scott A. Zinck

5aBA11. The effect of crystal chemical composition on the color Doppler ultrasound twinkling artifact
Student author: Eric Rokni

Session 3pBAb

Biomedical Acoustics, Signal Processing in Acoustics and Physical Acoustics: Interaction of Light and Ultrasound II

E. Carr Everbach, Cochair
Engineering, Swarthmore College, 500 College Avenue, Swarthmore, PA 19081

Ronald A. Roy, Cochair
Dept. of Engineering Science, University of Oxford, 30 Edgeway Road OX3 0HD, United Kingdom

Invited Papers

1:20

3pBAb1. Antivascular photo-mediated ultrasound therapy and its application in the eye. Xinmai Yang (Mech. Eng., Univ. of Kansas, 1530 West 15th St., Leaned Hall 3138, Lawrence, KS 66045, xmyang@ku.edu), Yannis M. Paulus (Dept. of Ophthalmology and Dept. of Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), and Xueding Wang (Dept. of Biomedical Eng. and Dept. of Radiology, Univ. of Michigan, Ann Arbor, MI)

Antivascular therapy can improve the prognosis of a variety of pathological conditions, including cancer and many eye diseases. By synergistically applying laser pulses and ultrasound bursts, we developed a photo-mediated ultrasound therapy (PUT) technique as a localized antivascular method. PUT takes advantage of the high native optical contrast among biological tissues and has the unique capability to self-target microvessels without damaging surrounding tissue. The technique utilizes an integrated therapeutic ultrasound and laser system. The laser system emits 5-ns, 10-Hz pulses, which is synchronized with 10-ms, 10-Hz ultrasound bursts. Experiments were carried out on chicken yolk sac membrane and rabbit eyes. With radiant exposures around 5 mJ/cm^{2}at 532 nm and ultrasound pressures around 0.4 MPa at 1 MHz or 0.5 MHz, microvessels were able to be removed. Furthermore, *ex vivo* tests with human blood demonstrated that cavitation was induced when laser and ultrasound were utilized synergistically. On the rabbit eye model, the blood flow in microvessels could be greatly reduced after PUT, and the occlusion of microvessels could last up to 4 weeks. In conclusion, PUT holds significant promises as a non-invasive method to precisely remove microvessels in neurovascular eye diseases by more selectively treating vasculature with minimized side-effects.

1:40

3pBAb2. Optical mean irradiance change for inference of focused-ultrasound-induced strain and thermal expansion in tissue-like materials. R. Glynn Holt (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, rgholt@bu.edu), Ali Vakili, Joseph L Hollmann, and Charles A. DiMarzio (Comput. and Elec. Eng., Northeastern Univ., Boston, MA)

Inferring material properties in scattering media, especially tissue, is a commonly applied method for detecting flaws or abnormalities. Hybrid acoustic/optic methods are often employed to overcome the limitations inherent in one or the other. Such hybrid methods have achieved great success in imaging various features (mechanical or optical properties) at a variety of spatial scales down to the subcellular. Here, we seek to develop an intermediate resolution method of detecting changes in material properties that is robust, inexpensive, and potentially capable of real-time analysis. We exploit the ability of the time-averaged absorption of focused ultrasound to induce changes in the optical index of scattering materials. By combining simulations with experiments, we demonstrate that the optical mean irradiance change measurement is capable of revealing time-dependent index changes. We are able to separate the contributions of both thermal expansion and radiation force deformation in the correlation signal. Potential applications of this technique will be discussed.

2:00

3pBAb3. Combining light and sound with nanoparticles to identify and treat head and neck cancers. Teklu Egnuni (Leeds Inst. of Medical Res., St. James' Univ. Hospital, Leeds, United Kingdom), Li Chunqi (School of Electron. and Elec. Eng., Univ. of Leeds, Leeds, United Kingdom), Nicola Ingram, Louise Coletta (Leeds Inst. of Medical Res., St. James' Univ. Hospital, Leeds, United Kingdom), Steven Freear, and James R. McLaughlan (School of Electron. and Elec. Eng., Univ. of Leeds, University of Leeds, Leeds LS2 9JT, United Kingdom, J.R.McLaughlan@leeds.ac.uk)

High intensity focused ultrasound (HIFU) is a non-invasive and non-ionising approach used primarily for the thermal ablation of cancerous tumours. Even though it has been in clinical use for over 30 years, it has yet to achieve widespread use. Two key limitations for this approach are long treatment times and a difficulty in getting real-time feedback on treatment efficacy. One technique that could help with these limitations is a combination of HIFU, pulse laser illumination, and cancer targeted nanoparticles. When nanoparticles are simultaneously exposed to these modalities, vapour bubbles form, providing a controllable way to nucleate cavitation in the target location. Acoustic emissions from inertial cavitation can be monitored via passive cavitation detection and/or mapping. This approach provides direct localisation of cancerous regions and has greater sensitivity compared with current photoacoustic imaging. Once the

cancerous regions have been localised, they can be ablated by HIFU, which is known to be enhanced in the presence of cavitation, by enhancing thermal damage in a localised region. Furthermore, the acoustic emissions generated during these ablations could give an indication of treatment progress. This study will present data on both *in vitro* and *in vivo* validation of this approach in models of head and neck cancer.

Contributed Papers

2:20

3pBAb4. HIFU tissue lesion quantification by optical coherence tomography. Jason L. Raymond (Dept. of Eng. Sci., Univ. of Oxford, 17 Parks Rd., Oxford OX1 3PJ, United Kingdom, jason.raymond@eng.ox.ac.uk), E. Carr Everbach (Eng., Swarthmore College, Swarthmore, PA), Ronald Roy (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Manuel Marques, Michael Hughes, and Adrian Podoleanu (School of Physical Sci., Univ. of Kent, Canterbury, Kent, United Kingdom)

Heating of tissue by high-intensity focused ultrasound (HIFU) can result in sufficient temperature elevation to cause irreversible changes in the tissue structure. The contiguous volume occupied by these changes, a lesion, and the extent of the tissue changes may be quantified histologically or estimated through techniques such as ultrasonic elastography. We have shown that changes in tissue optical scattering could be used as a proxy to improve sensing and imaging of HIFU lesion formation as an alternative to thermometry. Optical coherence tomography (OCT) is a light-based method appropriate for optically accessible tissues, which we have used to quantify lesion volume, shape, and quality based upon the irreversible changes in optical scattering that occurs with protein denaturation. We have adapted OCT to take into account changes in optical polarization of the tissue, providing sensitivity to changes in the collagen orientation of skin with heating. This technique has potential in detecting antecedents of skin burn during HIFU exposures, thereby increasing safety and reducing treatment times.

2:35

3pBAb5. Evaluation of a potential medical diagnosis application of sonoluminescence. Alicia Casacchia (Walker Dept. Mech. Eng., Univ. of Texas at Austin, 204 E. Dean Keeton St., Austin, TX 78712, acasacchia@utexas.edu), Parker George (Plan II Honors Program, Univ. of Texas at Austin, Austin, TX), Preston S. Wilson, and Mark F. Hamilton (Walker Dept. Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Sonoluminescence (SL) is a phenomenon in which light is produced via violent collapse of a gas-filled cavity under excitation by a varying pressure field. Although the precise mechanism of this light production is not yet agreed upon in the literature, various applications of this effect have been established and are continually being developed. One such example of an application of this phenomenon that has not been extensively studied is the production of SL in biological fluids as a method for medical diagnostics. Measurements performed by Chernov *et al.* [in Proceedings of 14th International Symposium on Nonlinear Acoustics (1996), pp. 219–223] revealed varied intensities of SL emissions in blood plasma samples from groups of patients diagnosed with different diseases. We present an experimental apparatus for the production of single bubble sonoluminescence (SBSL) and subsequent measurement of radial oscillations using optical scattering techniques. This system will allow for characterization of the effects of both the biological fluid content and viscoelasticity on the spectra of SBSL light emissions. Additionally, these experimental results can be used to inform future computational models of this behavior. [This is a Plan II SAWIAGOS project.]

2:50

3pBAb6. Deep-learning framework and acoustic reflector for improved limited-view and sparse photoacoustic tomography. Irvane Ngnie Kamga, Steven Guan, and Parag V. Chitnis (BioEng., George Mason Univ., 12300 Oak CreenREEK Ln., Apt. 1009, Fairfax, VA 22033, ingnieka@gmu.edu)

Photoacoustic imaging is a hybrid imaging modality that relies upon optical absorption of pulsed light and subsequent thermoelastic generation of ultrasound. The detection of the induced acoustic waves outside the tissue enables image reconstruction. A major challenge encountered in photoacoustic tomography (PAT) lies in the inability to acquire complete projection data from the region of interest due to both the limited view and sparsity of available ultrasonic sensors. The resulting images are characterized by severe artifacts and poor quality. In this work, we examined the utility of incorporating an acoustic reflector to address the limited view problem and to train a convolutional neural network (CNN) to improve PAT image reconstruction from sparsely sampled data. Photoacoustic wave propagation was simulated in MATLAB using the k-Wave toolbox. We compared the performance of a sparse linear transducer array (with and without reflector) to that of a circular transducer array. The structural similarity index (SSI) was used as a metric for evaluating image quality. The combination of a curved reflector and artifact-removal using a CNN improved the quality of PAT images from the linear configuration. The resulting mean SSI value (0.859) was comparable to that achieved using the circular transducer array (0.926).

3:05

3pBAb7. Pixel-wise deep learning for improving image reconstruction in photoacoustic tomography. Steven Guan, Amir Khan, Siddartha Sikdar, and Parag V. Chitnis (BioEng., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, Sguan2@gmu.edu)

Photoacoustic tomography involves absorption of pulsed light and subsequent generation of ultrasound, which when detected using an array of sensors can produce clinically useful images. Practical considerations limit the number of sensors and their "view" of the region of interest (ROI), which can result in significant reconstruction artifacts. Iterative-reconstruction methods can improve image quality but are computationally expensive. Another approach to improve reconstructed images is to use convolution neural networks (CNN) as a post-processing step for removing artifacts. However, missing or heavily obscured features typically cannot be recovered using this approach. We present a new pixel-wise deep learning (PDL) approach that employs pixel-wise interpolation to window ROI-specific raw photoacoustic data and then directly performs the image reconstruction within the CNN framework. The utility of this approach was demonstrated on simulated photoacoustic data from a 64-element semi-circular sensor array. The training and testing datasets comprised of 500 images from a synthetic vasculature phantom and 50 images of an anatomically realistic vasculature obtained from micro-CT images, respectively. The structural similarity index of the PDL-reconstructed images (0.91 ± 0.03) indicated superior image quality compared to those obtained using the iterative reconstruction (0.82 ± 0.09) and CNN-based artifact removal (0.79 ± 0.07).

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Christina J. Naify, Chair
Acoustics, Jet Propulsion Lab, 4800 Oak Grove Dr, Pasadena, CA 91109

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Hot topics in computational acoustics. D. K. Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr.-CRREL, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil)

As reflected by the formation of a new Computational Acoustics Technical Specialty Group in the ASA, computational methods attract acoustics researchers and practitioners across the spectrum of the current ASA Technical Committees. This presentation samples several prominent hot topics in computational acoustics: (1) high-resolution, three-dimensional solutions for sound fields in the ocean, atmosphere, and indoor spaces, (2) time-domain treatments of dissipative processes and reactive boundary conditions, (3) machine learning and other data-driven surrogate models trained with more computationally intensive physics-based models, and (4) characterization of uncertainty in computations and its quantification through methods such as adaptive sampling and polynomial chaos.

1:35

3pID2. Acoustic metamaterials and additive manufacturing. Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

Acoustic metamaterials (AMM) have captured the attention of a wide range of researchers for their potential to create materials with properties or functionality that exceed naturally occurring materials or conventional composites. Progress in acoustic metamaterials is largely the product of a combination of technological advances that have been made in the past three decades. Specifically, AMM research is a prototypical example of novel concepts in physics converging with advances in technology, primarily additive manufacturing and widespread access to robust computational tools. Additive manufacturing is a key component to this area of research since it allows for rapid build and test cycles as well as the production of elaborate structures for acoustic wave manipulation that follow from rigorous mathematical predictions. This talk will highlight physical concepts central to AMM research and the role that advanced manufacturing processes play in present and future research. It will then discuss the significant technical challenges that must be overcome if AMM are to be brought to their full potential, including the need to fabricate large amounts of subwavelength dynamic microstructure spanning orders of magnitude in length scale.

2:05

3pID3. Emerging imaging methods in biomedical ultrasound. T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, 3938 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Possibilities for diagnostic and therapeutic ultrasound imaging methods are continually extended by advances in transducer and beamformer technology, algorithms for signal processing and image reconstruction, research-oriented imaging platforms, and computing power. This talk describes and illustrates recent advances in biomedical ultrasound imaging methods from multiple groups. In different respects, each provides information beyond the tissue reflectivity, blood flow, and strain mapped by current clinical ultrasound scanners. Methods discussed include quantitative imaging of tissue parameters by solving inverse scattering and inverse elasticity problems, super-resolution imaging beyond the diffraction limit, and passive imaging of acoustic cavitation. Physical acoustics principles underlying the methods, as well as applications in diagnosis and therapy guidance, are discussed.

Session 3pMU

Musical Acoustics, Psychological and Physiological Acoustics and Signal Processing in Acoustics: Polyphonic Pitch Perception and Analysis II

Jonas Braasch, Cochair
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M. Torben Pastore, Cochair
Architectural Acoustics, Rensselaer Polytechnic Institute, 4 Irving Place, Troy, NY 12180

Invited Papers

1:00

3pMU1. Monaural edge pitch and models of pitch perception. William M. Hartmann (Phys. and Astronomy, Michigan State Univ., Physics-Astronomy, 567 Wilson Rd., East Lansing, MI 48824, hartmann@pa.msu.edu), Peter Cariani (Hearing Res. Ctr., Biomedical Eng., Boston Univ., Newton, MA), and H. Steven Colburn (Hearing Res. Ctr., Biomedical Eng., Boston Univ., Boston, MA)

Noise with a sharp spectral edge produces a definite, perceived pitch. The pitch of lowpass noise lies slightly below the edge and the pitch of highpass noise slightly above it. The pitch shifts away from the edge, expressed in semitones, increase with the decreasing edge frequency. A neural timing model based on positive peaks of the autocorrelation function accounts for the broad features of the shifts. So, does a place model based on the peak of an excitation pattern as enhanced by lateral inhibition. As the edge frequency decreases below 150 Hz, the pitch persists for a lowpass noise but disappears for highpass noise. This result is consistent with the timing model but not with the place model. For high edge frequencies, the pitch is stronger for highpass noise than for lowpass noise—consistent with both timing and place models. As the edge frequency approaches 5000 Hz, the pitch disappears for most listeners, but for some listeners, a pitch persists for edge frequencies beyond 8000 Hz. The latter result argues against a timing model. It appears that both timing processes (low edge frequency) and place processes (high edge frequency) are required to explain the edge pitch. [Work supported by the AFOSR and the NIDCD.]

1:20

3pMU2. Decoding MEG responses to musical pitch reveals the dynamic emergence of tonal structure in human cortex. Narayan Sankaran (Neurological Surgery, Univ. of California, San Francisco, 675 Nelson rising Ln., San Francisco, CA 94158, narayan.sankaran@ucsf.edu), Thomas A. Carlson (Psych., The Univ. of Sydney, Sydney, NSW, Australia), and William Forde Thompson (Psych., Macquarie Univ., Sydney, NSW, Australia)

Tonal music is characterized by a hierarchical structuring of pitch, whereby certain tones appear stable and others unstable within their musical context. Despite its prevalence, the cortical mechanisms supporting such a percept remain poorly understood. We examined the neural processing dynamics underlying pitch-structure in Western Tonal Music. Listeners were presented with tones embedded within a musical context whilst their magnetoencephalographic (MEG) activity was recorded. Using multivariate pattern analysis, decoders attempted to classify the identity of tones from their corresponding MEG activity at each peristimulus time-sample, providing a dynamic measure of their cortical dissimilarity. Time-evolving neural distinctions were then compared with the predictions of several acoustic and perceptual models. Following the onset, a temporal evolution was witnessed in the representational structure in cortex. While MEG dissimilarities between tones initially corresponded to their fundamental frequency separation, distinctions beyond 200 ms reflected their status within the hierarchy of perceived stability. Transposing dissimilarities corresponding to this latter period into different keys, neural relations between keys correlated with the well-known *circle of fifths*. Convergent with fundamental principles of music-theory and perception, results detail the dynamics with which the complex perceptual structure of Western tonal music emerges in human cortex within the timescale of an individual tone.

Contributed Paper

1:40

3pMU3. Automated Mandarin tone classification using deep neural networks trained on a large speech dataset. Jeramey Tyler, Huang Zou (Cognit. Sci., Rensselaer Polytechnic Inst., 1709 Tibbits Ave., Troy, NY 12180, tylerj2@rpi.edu), Helen Zhou (Commun. and Media, Rensselaer Polytechnic Inst., Troy, NY), Hui Su (IBM, Troy, New York), and Jonas Braasch (Rensselaer Polytechnic Inst., Troy, NY)

A tonal language is one in which the speaker's intonation modifies the meaning of a word. In this work, we perform a rigorous analysis of intonation changes, or pitch contours, produced by native Mandarin speakers to predict the tone-contour type. Pitch contours are estimated using a number of different methods, also measuring each contour's Mel-Frequency Cepstral Coefficients (MFCCs). The dataset used was autonomously generated from the Aishell open-source Mandarin speech corpus. Each sample was aligned with its transcript using Montreal Forced Alignment and segmented into individual words. The resulting corpus covers 11 topic domains, spoken by 400 individuals. Separate development, training, and testing datasets are created to ensure the integrity of our results. Pitch contours and their MFCCs are exposed to a number of machine learning techniques including clustered, regression, and traditional Deep Neural Network (DNN) approaches. MFCCs are additionally processed using convolutional neural networks. The models are used to predict the corresponding tone for a contour. Our work seeks to determine which intonation representations perform optimally for machine learning tasks. The tool is used to provide audio and visual feedback to learners of tonal languages. [Work supported by RPI Seed Grant and CISL].

Invited Paper

1:55

3pMU4. Multiple f0 pitch estimation for musical applications using dynamic Bayesian networks and learned priors. David A. Dahlbom and Jonas Braasch (Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, dahlbd@rpi.edu)

The identification of multiple simultaneous pitches is a challenging signal processing task and cannot at present be performed as well as trained human subjects. Moreover, it appears that successful human performance depends on skill acquisition and knowledge of musical conventions. Even human capabilities are likely fairly poor in the absence of training and musical context. We present a framework, using Dynamic Bayesian Networks, that permits the principled incorporation of models of music theory, musical instruments, and human pitch perception. A particular advantage of this approach is the ability to develop each of these models independently, relying either on expert knowledge or machine learning as necessary. Models of appropriate complexity can then be selected for a specific application. In the present work, we focus on the use of learned models of musical context, specifically Deep Markov Models, and use these to improve inferences about simultaneous pitches. The main drawback of this framework is the intractability of the inference computations and the computational expense of approximation methods. We explore particle filtering as an approach to addressing these problems with the ultimate aim of making a system useable in a musical performance system. [Work supported by NSF BCS-1539276 and IBM AIRC grant.]

Session 3pNS

Noise, Architectural Acoustics, and Structural Acoustics and Vibration: Noise at Sporting Events and Sports Venues

Daniel A. Russell, Cochair
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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—1:20

Invited Papers

1:25

3pNS1. Impulsive exposures from ceremonial and signal cannons. Gregory Flamme (SASRAC, 2264 Heather Way, Forest Grove, OR 97116, gflamme@sasrac.com), William J. Murphy, Chucri A. Kardous, and David C. Byrne (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH)

Cannons, small firearms, and starter pistols are sometimes used with blank charges during sporting events, ceremonies, and historical re-enactments. The sound levels produced by such devices are not widely known, and it is possible that the personnel discharging them could underestimate the potential risk to hearing. Depending upon the proximity to participants and spectators, the sound levels produced by such devices can be potentially hazardous. The exposures have not been widely examined because they fall outside of regulations and standards that cover typical occupational or military exposures. This presentation describes the acoustic characteristics and exposure limits for two large-caliber ceremonial cannons and a signal cannon. The cannons produced impulses between 150 and 174 dB peak sound pressure level (SPL) and 8-h equivalent A-weighted levels ranging between 60 and 108 dBA, respectively. In addition, measurements from small firearms and starter pistols are presented, which produced sound levels between 145 and 165 dB SPL. Such sound levels exceed the various occupational recommended and permissible exposure limits. This paper provides recommendations for noise and administrative controls for all personnel within 15 m of the ceremonial cannons and 10 m of the signal cannon. Double hearing protection should be required during all activities.

1:45

3pNS2. Referee whistles Part I—Permissible exposures indoors. Trevor W. Jerome (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, MS 3220B, State College, PA 16804-0030, twjerome@gmail.com), Gregory Flamme (SASRAC, Forest Grove, OR), and William J. Murphy (Div. of Appl. Res. and Technol., National Inst. for Occupational Safety and Health, Cincinnati, OH)

Sound from referee whistles at sporting events is usually relatively short in duration (<250 ms) but generated at relatively high levels (>110 dB SPL). Damage risk criteria (DRC) that categorize potentially harmful sounds are usually meant for either continuous or impulsive noise. These types of whistle sounds are better categorized as impulsive. Measurements were taken of a trained referee using a sample of commercially available whistles in a controlled environment. One microphone was placed at the ear of the referee, and another was placed 1 m in front of the referee. Whistle signals were analyzed with a maximum duration of 450 ms, which would be typical of a sporting event. DRC for these impulsive sounds have been investigated using DRC of A-weighted 8-h energy (LeqA8), and the warned and unwarned conditions of the AHAAH model. Depending on the risk model used, the numbers of permissible exposures was as low as zero and extended above.

2:05

3pNS3. Referee whistles Part II—Outdoor sound power assessment. William J. Murphy (Hearing Loss Prevention Team, Ctr. 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), Stephen M. Tasko (Speech Pathol. & Audiol., Western Michigan Univ., Kalamazoo, MI), Donald Finan, Deanna K. Meinke (Audiol. & Speech-Lang. Sci., Univ. of Northern Colorado, Greeley, CO), Michael Stewart (Dept. of Commun. Disord., Central Michigan Univ., Mount Pleasant, MI), James E. Lankford (School of Allied Health and Communicative Disord., Northern Illinois Univ., Dekalb, IL), Adam R. Campbell (Hearing Loss Prevention Team, Ctr. for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH), and Gregory Flamme (SASRAC, Forest Grove, OR)

Referee whistles have been suggested as a significant contributor to noise-induced hearing loss. Thirteen models of sport whistles were tested for sound power with a 3-m hemispherical array of 19 microphones. The whistler produced nine tweets of low, medium, and high effort with two samples of each whistle model. Sound power levels ranged between 74 and 115 dB re 1 pW. The low, medium, and high effort tweets had average power levels of 85 $\pm$ 6 dB, 100 $\pm$ 6 dB, and 110 $\pm$ 4 dB, respectively. Preliminary damage-risk analysis of the whistle impulses yield varied estimates for the allowable number of tweets before auditory damage might be expected. For the Auditory Hazard Assessment Algorithm, between 4 and 66 tweets may exceed the daily exposure threshold. Based upon the amount of eight-hour equivalent A-weighted energy, approximately 120 to 500 tweets would exceed the daily 85 dBA exposure limit. The directivity of the sound power measurements will also be examined, and risk of hearing loss will be discussed.

2:25

3pNS4. Improved automated classification of basketball crowd noise. Mylan R. Cook, Brooks A. Butler, Katrina Pedersen, Spencer Wadsworth, Eric Todd, Kent L. Gee, Mark K. Transtrum, and Sean Warnick (Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com)

This paper describes using both supervised and unsupervised machine learning (ML) methods to improve automatic classification of crowd responses to events at collegiate basketball games. This work builds on recent investigations by the research team where the two ML approaches were treated separately. In one case, crowd response events (cheers, applause, etc.) were manually labeled, and then, a subset of the labeled events were used as a training set for supervised-ML event classification. In the other, (unsupervised) k-means clustering was used to divide a game's one-twelfth octave spectrogram into six distinct clusters. A comparison of the two approaches shows that the manually labeled crowd responses are grouped into only one or two of the six unsupervised clusters. This paper describes how the supervised ML labels guide improvements to the k-means clustering analysis, such as determining which additional audio features are required as inputs and how both approaches can be used in tandem to improve automated classification of crowd noise at basketball games.

Session 3pPA

Physical Acoustics, Engineering Acoustics, and Biomedical Acoustics: Acoustofluidics II

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Charles Thompson, Cochair
ECE, UMASS, 1 University Ave., Lowell, MA 01854

Chair's Introduction—1:00

Contributed Papers

1:05

3pPA1. Simultaneous trapping and pulling by acoustical Bessel beams as stable tractor beams. Xudong Fan and Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, University, Oxford, MS 38677, xfan1@go.olemiss.edu)

A stable tractor beam for long range pulling of particles by acoustic Bessel beams requires a simultaneous trapping in the lateral direction. Trapping force acting on a sphere centered on the axis of Bessel beams is examined to guide the selection of material and beam parameters. It is found that a heavy and rigid sphere in the Rayleigh regime can be trapped at central pressure maximum of zero-order Bessel beam, which is contrary to the repelling behavior at the pressure anti-nodes of one- or two-dimensional standing waves. The trapping results from the three-dimensional features of the velocity fields and the momentum projection. The projection leads the trapping of an elastic sphere or a droplet at the axis of ordinary and vortex Bessel beams in the Rayleigh regime to be reversed when reducing the conical angle of the beam at the situation when there is a strong contrast of mass density between the particle and the surrounding fluid. Ranges of conical angle and material parameters for simultaneous trapping and pulling a spherical object are identified.

1:20

3pPA2. Acoustic streaming in a channel a moderate streaming Reynolds number. Charles Thompson (ECE, UMASS, 1 University Ave., Lowell, MA 01854, charles_thompson@uml.edu), Jairo Vanegas (ME, UMASS Lowell, Lowell, MA), Russell Perkins, Flore Norceide, Ivette Alvarez, and Kavitha Chandra (ECE, UMASS, Lowell, MA)

In this work, the generation of acoustic streaming in a rigid walled channel is examined. At low values of the streaming Reynolds number, the time-averaged fluid motion in the channel follows that given by Rayleigh. However, departure from the aforementioned result ensues as the magnitude of the streaming Reynolds number increases. Higher order nonlinear corrections to the Rayleigh streaming solution is given and are expressed in terms of a regular perturbation sequence in nondimensional particle amplitude. It is shown that the reduction in the amplitude of the axially directed streaming velocity is a function of the streaming Reynolds number.

1:35

3pPA3. Numerical modeling of submicron particle acoustic concentration in gaseous flow. Jizhou Liu (School of Energy and Power Eng., Beihang Univ., No. 37 Xueyuan Rd., Beijing 100191, China, jizhou.liu@buaa.edu.cn), Xiaodong Li (School of Energy and Power Eng., Beihang Univ., Beijing, China), and Fang Q. Hu (Mathematics and Statistics, Old Dominion Univ., Norfolk, VA)

The acoustic concentration of submicron particles in a micro-channel is investigated numerically via a gas-particle multiphase coupling scheme. For modeling the gas flow under transverse standing wave, 2 dimensional linearized Euler equations with the parabolic mean flow are employed with the high order finite difference method. Through the analogous behavior of rarefied gas and air-suspended particles, a modified Unified Gas-Kinetic Scheme (UGKS) is adopted to estimate the particle dynamics. In detail, Stokes' drag force and acoustic radiation force applied on particles are accounted for by introducing a velocity-dependent kinetic acceleration term in the UGKS. To validate this modeling, numerical simulations are tested with varying standing wave amplitudes. The effects of acoustic radiation force to drag force ratio and mean flow velocity are also analyzed. The computed concentration patterns and efficiencies are compared with experimental results from the literature. The agreement shows that the adopted Euler-UGKS coupling scheme could be an effective tool for micro-sized particle acoustic concentration applications.

1:50

3pPA4. An acoustics separation technique based on the development of an interface in the acoustic field. Krishna N. Kumar, Adrian Barber, Jack Saloio, Tyler Campbell, Kedar C. Chitale, Benjamin P. Ross-Johnsrud (Res. & Development, FloDesign Sonics, Inc., 380 MainSt., Wilbraham, MA 01095, k.kumar@fdsonics.com), and Bart Lipkens (Res. & Development, FloDesign Sonics, Inc., Springfield, MA)

Chimeric antigen receptor (CAR) T-cell therapy is a promising and evolving immunotherapy approach for cancer treatment. In allogeneic CAR-T therapies, TCR + cells must be removed from the final cell product because of immunogenicity problems. It is accomplished through a negative affinity cell selection process where TCR + cells are affinity bound to a

bead. The harvested TCR-cells are the product cells. A multidimensional acoustic standing wave field separates cell-bead complexes from free cells in an acoustic fluidized bed. The feed solution motion is normal to the primary acoustic field. Irrespective of the particle acoustic contrast, an interface between a dense suspension on the bottom and clear fluid on top develops in the field. We examine the physics behind the development of the interface and its subsequent motion. This motion influences the purity, scalability, and recovery of the TCR- cells. We present the effects of different acoustofluidic parameters, e.g., bead concentration, bead acoustic contrast factor, frequency, and flow rate on interface formation and its movement. Theoretical calculations and experimental results are discussed. The acoustic fluidized bed has been shown to give final purities of 99 + % of TCR- cells from a starting purity of 60%–70%, with 70 + % recoveries of TCR- cells.

2:05–2:20 Break

2:20

3pPA5. Droplet extraction and manipulation at a fluid interface using fraxicon modified ultrasound. Robert L. Lirette (Phys., Univ. of Mississipi, 2400 Anderson Rd., Apt. 4, Oxford, MS 38655, rlirette@go.olemiss.edu), Joel Mobley (Phys., Univ. of Mississipi, University, MS), and Likun Zhang (Phys., Univ. of Mississipi, Oxford, MS)

Ultrasound focused at a fluid-fluid boundary creates an acoustic radiation pressure on the boundary that is dependent on the incident energy density and the relative density and sound speed of each fluid. For different fluid combinations, this radiation pressure can either be positive or negative. For this study, ultrasound propagating from water to carbon tetrachloride was used to create a negative radiation pressure at the interface. This fluid combination is impedance matched eliminating reflections and heating effects at the boundary. A fraxicon phase plate lens is a low profile analog of an axicon and generates an approximate Bessel beam in the far field. The near field exhibits a complex diffraction pattern including shadow zones capable of acoustic trapping. Starting with a planar interface, we demonstrate the extraction, capture, and manipulation of a carbon tetrachloride droplet. The negative radiation pressure draws the carbon tetrachloride surface up into the water, eventually breaking a droplet free. The trapped droplet is then transported through the water by moving the transducer.

2:35

3pPA6. Development of heavy metal ions detector driven by surface acoustic waves. Yue Liu, Chaohui Wang, and TengFei Zheng (School of Mech. Eng., Xi'an Jiaotong Univ., No. 28, Xianning West Rd., Xi'an 710049, Shaanxi, China , 492865529@qq.com)

Environmental pollution caused by heavy metals is a global problem. Most of the metal ions lead to serious health concerns by producing free radicals. Therefore, rapid and accurate detection of metal ions has become an urgent problem. We develop a highly sensitive detector to detect heavy metal ions driven by surface acoustic waves (SAWs). The detector consists of a micro-reaction cell, a three-electrode system for electrochemical measurements, and two focused interdigital transducers (FIDTs) that produce SAWs. The SAWs promote liquid mixing effectively by sufficiently stirring the aqueous solution; therefore, the oxidation current in electrochemical reactions is enhanced. Furthermore, the center angles of the FIDT and the distance between two FIDTs turn out to be critical parameters in reaction efficiencies. These two factors can affect the SAW radiation areas which can influence the reaction efficiencies. This study realizes rapidly and accurately detecting heavy metal ions and provide a basis for designing SAWs-based detector for different chemical reactions.

2:50

3pPA7. Standing surface acoustic waves controlling crystal shapes: The case of urea. TengFei Zheng, Yue Liu, and Chaohui Wang (School of Mech. Eng., Xi'an JIiaotong Univ., No. 28, Xianning West Rd., Xi'an 710049, Shaanxi, China, 492865529@qq.com)

It is important to control crystal shapes, for this is particularly significant in nano-technology and in medicine. In this paper, we use standing surface acoustic waves (SSAWs) based technique to control the urea crystal shapes in a microdroplet. The surface acoustic waves are excited by a sinusoidal electric potential applied to inter-digital transducers on the surface of a X-propagating lithium niobate (LN) single-crystal piezoelectric substrate. A microdroplet is placed on the waves propagation path. SSAWs radiate into the droplet and influence the crystallization process when the droplet evaporates. First, we investigated the effect of acoustic intensity. Second, we controlled the shape of the crystal by changing the wave frequency. Finally, we controlled the shape of the crystal by changing the LN surface properties. Moreover, the streaming induced by SSAWs was observed by particle image velocimetry. Moreover, it is easier to control the wave frequency, acoustic intensity, and the surface properties than to control acoustic cavitation. This research will promote the application of SSAWs in controlling crystal forms.

3:05

3pPA8. Overview of acoustical levitation for volumetric visual displays. Carl Andersson and Jens Ahrens (Chalmers Univ. of Technol., Sven Hultins Gata 8a, Gothenburg 412 58, Sweden, carl.andersson@chalmers.se)

Acoustical levitation is an area with many applications ranging from medical drug delivery to micro-particle sorting. An application which has gained attention recently is the creation of volumetric displays by using small levitating objects. Advances in inexpensive ultrasonic phased arrays have increased the availability of dynamically controllable beamformers which enables the manipulation of the levitating objects in time and space. This allows for interpreting the levitating objects similarly to pixels in a normal display yet with an additional spatial dimension. Most implementations so far are based on the so-called Gor'kov formulation of radiation force. This formulation coupled with numerical optimization allows for calculation of the phases of the individual elements in the array. By exploiting symmetries in the solution, it is possible to impose an acoustic trap signature phase pattern onto simple focusing methods. Using off-the-shelf mid-air ultrasonic haptics systems to provide multiple focus points to which the phase patterns are applied allows for real-time control of multiple levitating objects. Current systems are limited to a handful of individually controllable objects so visualization is limited to abstract information. We present an overview of the state-of-the-art and discuss limitations and possibilities. [Work supported by Horizon 2020, No. 737087.]

Session 3pPPa

Psychological and Physiological Acoustics, Speech Communication, and Education in Acoustics: Diversity in Auditory Perception and Speech Communication

Kelly L. Whiteford, Cochair
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Dominique A. Bouavichith, Cochair
Linguistics, University of Michigan, 611 Tappan St., #455C, Ann Arbor, MI 48109

Evelyn M. Hoglund, Cochair
Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43204

Chair's Introduction—1:30

Invited Papers

1:35

3pPPa1. Racial categorization and word identification: The influence of sex, race and regional dialect. Yolanda F. Holt (Commun. Sci. and Disord., East Carolina Univ., 300 Moye Bv 3310-X HSB, MS 668, Greenville, NC 27834, holty@ecu.edu) and Tessa Bent (Indiana Univ., Bloomington, IN)

The speech signal provides information on talker characteristics including socio-ethnic affiliation and racial identity. Regional variation, both similar and divergent from White American English, has been described in African American English. However, it is unknown if such regional dialect variation influences listeners' racial categorization or word identification accuracy. This work evaluated the influence of listeners' sex, race, and regional dialect on racial categorization and word identification for Black and White talkers from two dialect regions within North Carolina. Black and White listeners (n = 23) from eastern and central North Carolina participated. In the racial categorization task, listeners heard /hVd/ words produced by male and female Black and White talkers from eastern and western North Carolina. Listeners categorized the perceived talker race for each token as Black or White. In the word identification task, the same listeners matched the speech tokens from the same talkers to one of fourteen /hVd/ words. Results showed an effect of listener sex on word identification accuracy such that female listeners were more accurate than male listeners. No effect of listener race or regional dialect was observed for either task. Follow-up analyses will investigate the interaction between listener and talker sex, race, and regional dialect.

1:55

3pPPa2. The southern shift and regional identity in Appalachia. Paul E. Reed (Communicative Disord., Univ. of Alabama, 909 Welsh Humanities Bldg., University of South Carolina, Columbia, South Carolina 29208, reedpe@email.sc.edu)

The Southern Vowel Shift (SVS) historically occurred across the Southern U.S. (e.g., Labov *et al.*, 1972; Feagin, 1986). Several recent studies document the retreat of the SVS in the urban South (Fridland, 1999; Prichard, 2010; Dodsworth and Kohn, 2012). However, Irons (2007) found the SVS advancing in rural Kentucky. Thus, Fridland notes the SVS might not be a supra-regional norm, serving rather as an ecological distinction within the South (2012:187). As an ecological distinction, SVS features might reflect a speaker's orientation toward differing ecologies. Thus, a speaker with a rural orientation might use more SVS features than a speaker who is less rurally oriented. The present paper investigates the SVS in rural speech examining three features: monophthongization of /aI/, rotation of /E/ and /e/, and rotation of /I/ and /i/. Data come from sociolinguistic interviews, reading passages, and words lists from 25 speakers from rural northeast Tennessee. Additionally, every participant completed a Rootedness Metric, a psychometric survey that quantifies place-based orientation. Results indicate that speakers exhibit SVS features, with reversals of the relative front vowel positions and monophthongization of /aI/ in all positions (cf. Irons, 2007). However, more rooted speakers exhibited the most advanced SVS features. Thus, the central difference of SVS features may not merely be ecological, rather the speaker's relationship to the differing regional ecologies.

2:15

3pPPa3. Proactive neural processing of native and non-native speech. Fernando Llanos (Commun. Sci. and Disord., Univ. of Pittsburgh, 507 Edmond St., 6, Pittsburgh, PA 15224-2036, f.llanoslucas@gmail.com), Rachel Reetzke (Health, Univ. of California Davis, Austin, Texas), and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Pittsburgh, Pittsburgh, PA)

The impact of attention and language-experience in neural speech processing is typically assessed using sentences, words, and syllables that, when presented in isolation, may not engage the cortical speech network as well as more realistic continuous speech. Here, we explore the neuromodulatory effects of attention and language experience using continuous speech. We recorded electroencephalographic responses from native speakers of English and late Chinese-English bilinguals while listening to a story recorded in English. The story was mixed with a tone sequence, and listeners were instructed to focus either on the speech (attended speech condition) or the tone sequence (ignored speech condition). We used the multivariate temporal response function and the accuracy of a machine-learning based brain-to-speech decoder to quantify differences in cortical entrainment and speech-sound category processing. Our analyses revealed a more robust, context-independent neural encoding of speech-sound categories when they were attended and native. Interestingly, while cortical entrainment to speech was also enhanced by attention, the enhancement was stronger among non-native speakers. Our results suggest that while listeners can manage attention to improve the neural parsing of continuous speech via cortical entrainment, the benefits of attention for speech-sound category processing can be attenuated when the sounds are not native.

2:35

3pPPa4. Turn up the volume: Speech perception in noise for bilingual listeners. Erika Skoe (Dept. of Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., U-1085, Storrs, CT 06105, erika.skoe@uconn.edu)

Bilinguals, compared to monolinguals, have been reported to have stronger neural representation of the fundamental frequency (F0) of speech, as measured by the frequency-following response (FFR). In monolinguals, stronger FFRs to F0 have been associated with better speech perception in noise (SPIN), suggesting that bilinguals should outperform monolinguals on SPIN tests. However, the opposite is the case: bilinguals generally underperform monolinguals. To explain such findings, Krizman *et al*. (2017), proposed that the bilingual brain might *turn up the volume* on the neural representation of sound to compensate for reduced SPIN. The current work considers this possibility using a combination of FFR and SPIN data. The Revised Speech in Noise test was administered at signal to noise ratios (SNRs) of 0 and 3 dB to young adult monolinguals ($n = 17$) and early bilinguals ($n = 26$). Unlike the monolinguals, the bilinguals showed a drop in performance when the SNR dropped. Within the bilingual group, poorer SPIN performance correlated with stronger neural responses to F0, suggesting that sensorineural areas are recruited to increase the neural gain of the acoustic representation in a manner that inversely correlates with speech comprehension. These findings give new insight into the brain-behavioral relationships for the neural encoding of sound.

2:55

3pPPa5. Effects of aging on voice emotion recognition in cochlear implant users and normally hearing adults listening to spectrally degraded speech. Shauntelle Cannon and Monita Chatterjee (Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, Shauntelle.Cannon@boystown.org)

Voice emotion recognition declines with age in normally hearing (NH) adults. However, the effects of aging on voice emotion recognition in NH listeners with spectrally degraded stimuli are neither known nor the effects of aging on adult cochlear implant (CI) users. This study explored age-related effects on voice emotion recognition in NH adults listening to spectrally degraded speech and aimed to compare these changes to age-related changes in adult CI users' voice emotion recognition with unprocessed speech. Participants listened to 12 emotion-neutral sentences, each spoken with 5 emotions (happy, sad, scared, angry, and neutral) by a male and female talker. Participants identified which emotion they heard while listening to sentences that were either unprocessed or CI-simulated. Preliminary results indicate declines in overall percent correct scores and increased reaction time with both age and increasing spectral degradation in NH adults. Results also suggest age-related effects on percent correct scores and reaction times within the CI group alone. These results have important implications in the aging population of adults with NH and with CIs because limitations in the quality peer to peer interactions have been associated with a decrease in perceived quality of life. [Work supported by R01 DC014233 and P20 GM109023.]

Session 3pPPb

Psychological and Physiological Acoustics and Speech Communication: Context Effects in Speech Perception II (Poster Session)

Christian Stilp, Cochair
Psychological and Brain Sciences, University of Louisville, 308 Life Sciences Building, Louisville, KY 40292

Matthew Winn, Cochair
Speech & Hearing Sciences, University of Washington, 1417 NE 42nd St., Seattle, WA 98105

All posters will be on display, and all contributors will be at their posters from 1:30 p.m. to 3:30 p.m.

Contributed Papers

3pPPb1. Sentence perception in noise by hearing-aid users: Relations with syllable-constituent perception and the use of context. James D. Miller (Res., Commun. Disord. Technol., Inc., 3100 John Hinkle Pl, Ste. 107, Bloomington, IN 47408, jamdmill@indiana.edu), Charles S. Watson (Res., Commun. Disord. Technol., Inc., Bloomington, IN), Marjorie R. Leek (Res., VA Loma Linda Healthcare System, Loma Linda, CA), David J. Wark (Commun. Sci. and Disord., Univ. of Memphis, Memphis, TN), Pamela Souza (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), Sandra Gordon-Salant (Hearing and Speech Sci., Univ. of Maryland, College Park, MD), Jayne B. Ahlstrom, and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Masked sentence perception by hearing-aid users is strongly correlated with three variables: (1) the ability to hear phonetic details as estimated by the identification of syllable constituents in quiet or in noise; (2) the ability to use situational context that is extrinsic to the speech signal; and (3) the ability to use inherent context provided by the speech signal itself. These conclusions are supported by the performance of 57 hearing-aid users in the identification of 109 syllable constituents presented in a background of 12-talker babble and the identification of words in naturally spoken sentences presented in the same babble. A mathematical model is offered that allows calculation of an individual listener's sentence scores from estimates of context utilization and the ability to identify syllable constituents. When the identification accuracy of syllable constituents is greater than about 55%, individual differences in context utilization play a minor role in determining the sentence scores. As syllable constituent scores fall below 55%, individual differences in context utilization play an increasingly greater role in determining sentence scores. When a listener's syllable constituent score is above about 71% in quiet, the listeners score in quiet will above about 55% in noise. [Watson and Miller are shareholders in Communication Disorders Technology, Inc., and may profit from sales of the software used in this study.]

3pPPb2. The role of linguistic variability in the perception of voice cues. Floor Arts, Etienne Gaudrain, Terrin N. Tamati, and Deniz Baskent (Dept. of Otorhinolaryngology / Head and Neck Surgery, Univ. Medical Ctr. Groningen, Hanzeplein 1, Groningen, The Netherlands, f.arts@umcg.nl)

Talkers' voices play an important role in speech perception. Through voices, we identify individual talkers and can facilitate speech communication in challenging conditions (e.g., cocktail party situations). While previous research has suggested that several linguistic factors broadly influence talker perception, how these factors influence perception of the individual voice cues remains unclear. The current study investigated the role of linguistic variability in voice cue perception, specifically fundamental frequency (F0) and vocal-tract length (VTL). Just-Noticeable-Differences (JNDs) were obtained using a 3AFC adaptive paradigm. Effects of word status (words, nonwords), word characteristics (lexical frequency, neighborhood density), and nonword characteristics (phonotactic probability, neighborhood density) were examined. Results demonstrated that voice cue perception was influenced by linguistic variability. While overall similar for words and nonwords, F0 and VTL JNDs were affected by phonological information in nonwords, i.e., phonotactic probability, but not on lexical information, i.e., lexical frequency and neighborhood density. However, VTL JNDs varied less across linguistic conditions than F0 JNDs, suggesting different processing mechanisms for these voice cues. These findings provide better insight into the interaction of voice and linguistic information, which may also improve our understanding of speech perception processes in populations with limitations in voice perception, such as cochlear-implant listeners.

3pPPb3. Interactions between auditory quality, linguistic context, short-term memory, and speech recognition. Adam K. Bosen (Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, adam.bosen@boystown.org)

The ability to recognize speech depends on several factors, including the quality of the auditory input, linguistic context, and the listener's cognitive abilities. These factors correlate with speech recognition, but less consideration has been given to how these factors interact. Here, we present evidence from two studies indicating that interactions exist between quality, context, and short-term memory. In study one, we demonstrate that contextual expectations across stimulus sets can determine the relationship between auditory quality and short-term memory. Specifically, recall of lists of digits is not affected by noise-band vocoding, whereas vocoding impairs both item identification and recall for lists of single syllable words drawn from a large, untrained set. In study two, we demonstrate that correlations between digit list recall and PRESTO sentence recognition are strongest when auditory quality is poor, whereas correlations between digit and word list recall weaken with decreasing auditory quality. This finding suggests that auditory quality and semantic context moderate the relationship between memory and speech recognition. We conclude that incorporating experimental measures of auditory quality, short-term memory, and recognition of speech materials with different contexts will provide a clearer perspective on how these factors relate to speech recognition in listeners with hearing loss.

3pPPb4. Examining use of context in sentences: The recruitment of perceptual and cognitive-linguistic abilities in younger and older adults. Daniel Fogerty, Rachel E. Miller (Commun. Sci. and Disord., Univ. of South Carolina, 1224 Sumter St., Columbia, SC 29208, fogerty@sc.edu), Jayne B. Ahlstrom, and Judy R. Dubno (Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Auditory-verbal contextual information facilitates speech recognition. In everyday listening, perceptual and cognitive-linguistic abilities influence speech recognition, but the role of these abilities in contextual use remains unclear. To assess the use of context, younger and older adults listened to sentences in multitalker babble and identified the final words from either high- or low-predictability sentence frames; difference scores between the two conditions indicate use of context. We also assessed (1) speech processing abilities, including masking release, talker segregation, and auditory-verbal closure, (2) cognitive abilities, including memory, inhibition, and speed of processing, and (3) linguistic abilities, assessed through verbal fluency and vocabulary knowledge. Compared to younger adults, older adults had better contextual use and vocabulary knowledge but had poorer recognition of final words with increasing age. These measures, combined with thresholds and age, were entered as predictor variables to explain performance for high- and low-context sentences. Preliminary results indicate that vocabulary knowledge and auditory-verbal closure were associated with contextual use and final word recognition in low-context sentences, respectively. Final word recognition in high-context sentences involved a combination of abilities. Thus, recruitment of perceptual and cognitive-linguistic abilities depends, in part, on the availability of sentence context. [Work supported by NIH/NIDCD.]

3pPPb5. Earlier music biases subsequent musical instrument categorization. Joshua M. Lanning and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., University of Louisville, Louisville, KY 40292, jmlann01@louisville.edu)

Perception of sounds occurs in the context of surrounding sounds. When spectral properties differ between earlier (context) and later (target) sounds, categorization of later sounds becomes biased through *spectral contrast effects* (SCEs). Past research has shown SCEs to bias categorization of speech and music alike. Additionally, the magnitudes of SCEs are not all-or-none but vary continuously in both speech and music categorization. Recently, the natural spectral composition of (unfiltered) sentences biased speech categorization via SCEs (Stilp and Assgari, under review). Here, we tested whether natural (unfiltered) music would similarly bias categorization of French horn and tenor saxophone targets. Preceding contexts were either solo performances of the French horn or tenor saxophone (unfiltered) or a string quintet processed to emphasize frequencies in the horn or saxophone (filtered). Categorization was influenced by SCEs in both unfiltered and filtered conditions, with more "saxophone" responses following horn / horn-like contexts and vice versa. Since unfiltered music produced SCEs in musical instrument categorization, this extends their influence to everyday listening conditions.

3pPPb6. Attention versus adaptation in processing talker variability in speech. Sung-Joo Lim, Jessica Tin, Allen Qu, and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Auditory Neurosci. Lab., Boston Univ., 610 Commonwealth Ave., Boston, MA 02215-2422, sungj.m.lim@gmail.com)

Speech processing is slower when listening to speech from multiple talkers compared to one continuous talker. Prior studies of talker variability mostly compared listening to long blocks of a continuous talker versus blocks where talkers constantly switch. It is thus unclear whether differences in processing efficiency are better understood as facilitation from perceptual adaptation to a talker or as interference from the attentional disruption caused by abrupt talker switches. It is also unclear how processing a single talker's speech becomes more efficient with ongoing exposure. Here, we examined how speech processing speed depends on preceding exposure to a talker. Listeners performed a speeded word identification task, in which they heard each talker's speech for 2–7 consecutive trials before the talker switched. Word identification was slowest when the talker switched and was expedited after a single exposure to a talker. However, additional exposure to a talker did not further improve word identification speed. More frequent talker switches in the preceding speech also led to slower subsequent word identification. Our findings suggest that speech processing efficiency does not depend on perceptual adaptation to the preceding talker; rather, slower speech processing after a change in talker reflects the costs of attentional reorientation.

3pPPb7. More information is more information: Working memory and visual context. Joanna H. Lowenstein and Susan Nittrouer (Speech, Lang., and Hearing Sci., Univ. of Florida, 1225 Ctr. Dr., Rm. 2147, P.O. Box 100174, Gainesville, FL 32610, jlowenstein@phhp.ufl.edu)

Listeners with cochlear implants demonstrate diminished auditory-verbal working memory capacities, possibly due to a lack of durable codes in the memory buffer. Earlier studies suggest that the context provided by lip-read information should enhance those codes, with both the phonological form and dynamic nature of lip-read signals contributing to this facilitative effect. The purpose of this study was to test the hypothesis that lip-read signals would make uniquely beneficial contributions to the recognition of degraded speech. To test this hypothesis, three kinds of signals were used: unprocessed words, vocoded words, and nonspeech environmental sounds. Two kinds of visual enhancements were applied: (1) dynamic signals specifying the event that generated the signal or (2) pictures representing the object named or creating the signal. Eighty young adults with normal hearing were asked to recall order of eight-item lists in a closed-set format. All listeners heard lists in all three signal conditions (unprocessed, vocoded, environmental sounds) but half recalled order in each visual-enhancement condition. Adding lip-read information improved accuracy and eased cognitive demands for recall of vocoded words, but other visual information provided benefits as well, calling into question previous claims of the specialness of dynamic facial movements.

Session 3pSA

Structural Acoustics and Vibration, Engineering Acoustics, Noise, and Architectural Acoustics: Novel Damping Treatments

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

Benjamin Beck, Cochair
Applied Research Lab, Penn State University, The Pennsylvania State University, Applied Research Laboratory, P.O. Box 30, MS 300B, State College, PA 16804-0030

Hubert S. Hall, Cochair
Mechanical Engineering, The Catholic University of America, 9500 MacArthur Blvd., Bldg. 3, Rm. 252.05, Code 7310, West Bethesda, MD 20817

Invited Papers

1:00

3pSA1. Shaping shock transmission with lightweight elastomeric materials. Ryan L. Harne and Peter Vuyk (Mech. and Aerosp. Eng., The Ohio State Univ., 201 W 19th Ave., E540 Scott Lab, Columbus, OH 43210, harne.3@osu.edu)

Effective suppression of impulsive elastic waves requires the reduction of the transmitted shock pulse and the elongation of shock duration. Recent experimental studies with engineered, lightweight elastomeric materials suggest that these requirements are met to large extent. The materials capitalize upon the mesoscale geometry that is known to collapse in unique ways according to the internal geometric design and magnitude of the impact force. Yet, the relations among material design, collapse trend, and resulting shock mitigation remain unknown. This research seeks to shed light on the connections using digital image correlation techniques that uncover exact origins of energy distribution through mapping of local strain fields. With a sequence of controlled shock experiments, we first identify how the impact force magnitude governs the classification of shock mitigation capability of the materials. Then, the relative variations of such trends as tailored by the internal material geometry are examined. All together, the results illuminate the range of working conditions and material designs for which shock attenuation capability of the materials remains exceptional.

1:20

3pSA2. Broadband elastic black hole and vibration isolator based on the transformation method. Hexuan Gao and Zhihai Xiang (Dept. of Eng. Mech., Tsinghua Univ., N510b, Mengminwei Technol. Bldg., Beijing 100084, China, gaohx15@mails.tsinghua.edu.cn)

The transformation method has stimulated many interesting applications of manipulating electromagnetic and acoustic waves by using metamaterials, such as super-lens imaging and cloaking. These successes are mainly due to the form-invariant property of the Maxwell equations and acoustic equations. However, the similar progress in manipulating elastic waves is very slow, because the elastodynamic equations are not form-invariant. Here, we show that the expression of the elastodynamic potential energy can approximately retain its form under two restrictions: conformal mapping and using the material whose longitudinal wave velocity is much larger than the transverse wave velocity. Based on this finding, we use inhomogeneous isotropic material to design and fabricate an efficient 180-deg wave bender acting as vibration isolator, in which the incident elastic waves turn around and make little disturbance on the item being supported. In addition, an elastic black hole is designed in the same way, so that the elastic energy propagating near the black hole is mostly absorbed into the center and dissipated by the damping material.

Contributed Papers

1:40

3pSA3. Design optimization of three styles of acoustic black hole vibration absorbers. Cameron A. McCormick (Appl. Res. Lab, Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, cam634@psu.edu) and Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., State College, PA)

Structures whose thickness is tapered according to a power law exhibit the acoustic black hole (ABH) effect and can provide effective vibration absorption with a net reduction in mass. However, it remains unclear what constitutes the best design of ABH vibration absorbers, including how the power-law taper is implemented in practice. This talk will present a formal optimization study of three styles of one-dimensional ABH vibration absorbers. Each ABH is embedded in a simply supported beam, which is excited by a harmonic force at one end. A multi-objective approach is used to identify the set of ABH designs that optimally minimize the structure's vibration response and its overall mass. Results show that each style has a similar tradeoff between the two objectives but that the choice of how the

power-law taper is implemented can be significant. Finally, the optimal designs will be evaluated on other criteria that may be of importance in the practical implementation of such ABH vibration absorbers, including buckling load and sound radiation.

1:55

3pSA4. Experimental design for the accurate measurement of ultra-low damping of simple structures. Hubert S. Hall, James J. Dlubac, and Michael Kim (Signatures, Naval Surface Warfare Ctr. Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, hubert.hall@navy.mil)

As a means of validating numerical models, recent research has shown a need for methodology to measure the structural damping of very lightly damped structures (loss factor < 0.005) to a high level of accuracy. Traditional experimental methods of measuring structural damping must be altered to accurately capture the lightly damped response. Otherwise, inaccurate damping values will be calculated that are larger than those intrinsically found in the material/structure. This presentation focuses on experimental technique modifications required for frequency domain methods of measurement of ultra-low damped structures. First, the implications of less accurate capture of resonant peaks in frequency response functions will be explored. Typical short frame, uniform frequency spacing can result in large measurement errors with lightly damped test articles. Instead, long time histories, focused sine-based testing, zero-padding, and peak approximation methods should be utilized for improved accuracy. Additionally, the effects of isolation-related boundary conditions and instrumentation mass to measurement accuracy are explored and quantified.

2:10

3pSA5. Impact of viscoelastic damping properties on vibration response of scale models. Benjamin Beck (Appl. Res. Lab., Penn State Univ., The Penn State Univ., P.O. Box 30, MS 3200D, State College, PA 16804-0030, benbeck@psu.edu)

There are many industries that utilized scale models to test the behavior of systems before building full-size structures, such as aircraft, spacecraft, and marine vehicle design. While vibration and acoustic response may not be the only quantities of interest when testing scale models, scale models can be utilized to predict full-scale acoustic behavior. When damping treatments are to be applied to full-scale systems, there will also be a need to apply these treatments to the scale models. Typically, these damping treatments use viscoelastic materials due to their high loss factor. However, since the stiffness and loss factor of viscoelastic materials change significantly with frequency, viscoelastic damping treatments will not behave consistently between model- and full-scale. This work shows the impact of using the same viscoelastic damping treatment on scale models as designed for full-scale systems. As a test case, a model of a thin vibrating plate will have a constrained layer damper applied on one surface. The total surface velocity response of a full-scale and model-scale finite element plate with a viscoelastic constrained layer damper is compared. Methods of designing viscoelastic materials for scale-models will also be presented.

2:25

3pSA6. Laser Doppler vibrometer and bending wave based dynamic material characterization of organic aerogel panels. Max Miller (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, millem23@rpi.edu), Sadeq Malakooti (Dept. of Mech. Eng., Univ. of Texas at Dallas, Richardson, TX), Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), Nicholas Leventis (Dept. of Chemistry, Missouri Univ. of Sci. and Technol., Rolla, MO), and Hongbing Lu (Dept. of Mech. Eng., Univ. of Texas at Dallas, Richardson, TX)

To what extent aerogels are useful in noise and vibration control applications remains an open question. With the advent of low cost, facile synthesis, ductile, and purely polymeric aerogels, the prospects are alluring. Within the polyurea family of aerogels, multiple nanomorphologies exist with some being amenable to rheological paradigms. An inverse problem approach is applied to tease out material parameters with the aid of viscoelastic models and a pair of dynamic characterization techniques. A specialized transfer function driven, quasi-longitudinal wave eliciting, material characterization setup is refined and augmented with laser Doppler vibrometry. This effort seeks to improve repeatability and increase frequency limits, two factors whose absences plague many schemas. Intriguing properties previously uncovered with this method, such as broadband negative dynamic mass, are reexamined. In tandem, a bending wave excitation arrangement and characterization method is introduced. Low bending and quasi-longitudinal phase speeds in addition to high static and low dynamic moduli shed light on the enhanced noise and vibration control performance reported in the literature. This paper discusses the dynamic measurement method and possible applications which may benefit from this unique combination of properties.

2:40

3pSA7. Analysis of increased damping in arrays of attached resonators. John A. Sterling, Joseph F. Vignola (Mech. Eng., Catholic Univ. of America, 620 Michigan Ave., Washington, DC 20064, jsterling@gmail.com), Teresa J. Ryan (Mech. Eng., East Carolina Univ., Greenville, NC), and Sai T. Paruchuri (Mech. Eng., Virginia Polytechnic Inst. and State Univ., Blacksburg, VA)

Research has shown that arrays of small dynamic attachments on a larger, primary structure can be tuned to significantly alter the time and frequency response of the system. Such attachments can be used to increase the apparent damping of a primary structure by transferring energy into the substructure and dissipating the energy there. By selecting the properties of the attachments to obtain specific distributions of element mass and mode frequency, the response of the primary structure can be altered to obtain specific spectral outcomes. Unfortunately, small errors in the distributions of mass and stiffness of the individual attachments have been shown to produce a significant degradation of intended performance. Until recently, metals were the practical option that was easily manufactured to the tolerances required. New three-dimensional-printing advancements have made plastics available with sufficient tolerances. A serendipitous discovery was that the increased damping shown in the plastics decreased the system's sensitivity to error. Previous analysis assumed low damping in the attached resonators. This work will show analysis with moderate to high damping levels for the attachments. Here, we will show experimental results using different plastic materials with moderate dimensional error.

Session 3pSC

Speech Communication: Second Language Speakers and Listeners (Poster Session)

Susannah V. Levi, Chair
Communicative Sciences and Disorders, New York University, 665 Broadway, 9th floor, New York, NY 10012

All posters will be on display, and all authors will be at their posters from 1:30 p.m. to 3:30 p.m.

Contributed Papers

3pSC1. A generational acoustic analysis of Mexican Spanish and English vowels. Edwin D. Reyes Herrera (Linguist, Macalester College, 1600 Grand Ave., Saint Paul, MN 55105, ereyeshe@macalester.edu)

Multiple studies have suggested that the Spanish vowel system is stable, with little to no variation (Delattre, 1996; Quillis and Esgueva, 1983), while others have also noted variation in monolingual Spanish (Boyd-Bowman, 1952; Canellada and Vicente, 1960; Delforge, 2008; Lipski, 1990; Lope-Blanch, 1964; Marín Galvés, 1995; Matluck, 1952; Quilis, 1999; Quilis and Esgueva, 1983) and bilingual Spanish in the United States (Fought, 1999; Godinez and Maddieson, 1985; Konopka and Pierrehumbert, 2008; Konopka, 2011; Ronquest, 2013; Willis, 2005). This study analyzes the vowel quality and duration of Spanish and English vowels by Mexican and Mexican-American speakers across different sociolinguistic generations in the United States (Silva-Corvalan, 1994), which has been absent from previous literature. An examination of the different generations is needed to account for the differing levels of Spanish and English use in the United States, as well as to examine how production of speech may change across generations. Results indicate all generations exhibit significant unstressed vowel reduction, though there is variation across the different groups. The amount of centralization and reduction also increases for each subsequent generation. In addition, Spanish influence is evident in the production of both G1 and G3 groups, though there is also evidence for acquired English speech.

3pSC2. A comparison of training effects on non-native tone sandhi production between American English and Cantonese speakers. Bei LI (The Hong Kong Polytechnic Univ., AG511, Hung Hom, Hong Kong, 199122libei@gmail.com), Yike Yang, Si Chen (The Hong Kong Polytechnic Univ., Kowloon, Hong Kong), and Yunjuan He (Univ. of North Georgia, Hong Kong, Hong Kong)

Previous studies suggest that productions of Mandarin tone sandhi by both American English speakers and Cantonese speakers were perceived more native-like after a laboratory perceptual training, whereas little is known about the effects of tonal or non-tonal backgrounds. Ten Cantonese-speaking trainees and ten American English-speaking trainees matched in age and Mandarin proficiency were recruited to the pre- and post-training recording sessions. Elicited with audio and visual stimuli, participants naturally produced disyllabic real and wug words where the two Mandarin tone sandhi rules (T3 + T1/T2/T4 sandhi and T3 + T3 sandhi rules) should be applied. In total, 7680 sandhi syllables obtained from two sessions were perceptually evaluated by two phonetically trained Mandarin-speaking raters on a 101-point scale. Statistical results indicated that native tonal/non-tonal backgrounds influence Mandarin learners' improvement in the two sandhi rules differently. The Cantonese trainees outperformed the English trainees in the sandhi of T3 + T1/T2/T4 before training, and the two groups had statistically comparable performance after training, although both groups exhibited significant improvement. For the sandhi in T3 + T3, improvement occurred for the Cantonese trainees while not for the American trainees after training, suggesting that the successful learning of phonological T3 sandhi rule may require a tonal background.

3pSC3. Cross-linguistic pitch differences in bilingual speakers. Andrew Cheng (Dept. of Linguist,Univ. of California, Berkeley, Berkeley, CA 94720, andrewcheng@berkeley.edu)

Studies in cross-language pitch differences show that languages can fundamentally differ in average pitch range even when speaker differences are accounted for (e.g., Mennen *et al*., 2012). Such studies usually compare similar populations of monolingual speakers but not bilingual speakers of the languages in question. Bilingual development research, in turn, does not address average language pitch but rather segmental properties such as vowel formants. The current study fills the gap by examining the fundamental frequency (f0, or pitch) of both languages of English-Korean bilinguals. Twenty Americans of Korean descent (female = 13, average age = 24) recorded natural speech during a bilingual interview. Their speech was digitally recorded and transcribed, and f0 was analyzed using the inverse filter control method (Ueda *et al*., 2007). Average f0 per word, per language, and per subject was then calculated and compared across individuals and groups. Results showed that Korean has significantly higher f0 than English, regardless of gender or age. The implications for research in acoustics and speech communication include a greater consideration for bilingualism or bilingual modes of speech as a variable, as well as an open line of inquiry into further study of cross-language differences within the bilingual speaker, including as a result of code-switching.

3pSC4. Effects of perceptual training in Mandarin tone sandhi production. Si Chen (The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Shuwen Chen (The Chinese Univ. of Hong Kong, Hong Kong, Hong Kong), Yunjuan He (Univ. of North Georgia, Hong Kong, Hong Kong), Bei LI (The Hong Kong Polytechnic Univ., AG511, Hung Hom, Hong Kong, 199122libei@gmail.com), Yike Yang (The Hong Kong Polytechnic Univ., Kowloon, Hong Kong), and Ratree Wayland (Univ. of Florida, Gainesville, FL)

It is well-established that the production of non-native lexical tone poses a great challenge to adult L2 learners (Hao,2012; Chang and Yao, 2016; Mok *et al*., 2018). The production of tonal patterns on disyllabic words is more challenging for non-native speakers because additional computational and/or lexical mechanisms are involved in correctly applying the tone sandhi rules in languages like Mandarin (Chen *et al*., 2017). Previous speech training studies have shown that perceptual training could improve both perception and production of non-native tones in isolation (Wang *et al*., 1999, 2003; Wayland and Li, 2008). The current study aims to further examine if perceptual training promotes learning of tonal patterns on disyllabic words by examining the production of two Mandarin Tone sandhi rules—the third tone sandhi and half-third tone sandhi by Cantonese learners. Native Cantonese speakers were trained with an identification task and a same/different discrimination task with both real and wug words. Their pre- and post-

training production was compared with native speakers' production, and it showed that perceptual training can lead to more successful learning of Mandarin tone sandhi rules based on acoustic and statistical analyses of tonal contours.

3pSC5. Acoustic characteristics of voiceless English fricatives (/f θ s/) produced by native English and native Japanese speakers. Katsura Aoyama (Univ. of North Texas, 1155 Union Circle #305010, Denton, TX 76703, katsuraaoyama@gmail.com), James E. Flege (Univ. of Alabama at Birmingham, Tuscania, Italy), Reiko Akahane-Yamada (ATR Int., Seika cho, Kyoto, Japan), and Tsuneo Yamada (Open Univ., Chiba, Japan)

This study examined productions of three English voiceless fricatives (/f θ s/) produced by native Japanese (NJ) and native English (NE) adults and children (16 participants each in 4 groups). The purpose of this study was to investigate acoustic characteristics of these fricative productions that were evaluated using intelligibility ratings in Aoyama *et al*. (2008). The following acoustic parameters were selected based on their importance in differentiating English fricatives (Jongman *et al*., 2000): noise duration, spectral moments, normalized amplitude, and relative amplitude. A total of 768 tokens were acoustically analyzed (256 each of /f θ s/). The results showed that there are many differences in acoustic characteristics between NJ and NE speakers. First, the durations of /θ/ were longer in the NJ speakers' productions than in the NE speakers' productions on average. Second, the spectral mean for /s/ was lower in the NJ speakers' productions than NE speakers' productions. Third, the NJ speakers' productions of /s/ did not differ from /f/ and /θ/ in normalized amplitude, whereas the NE speakers' productions of /s/ showed higher amplitude than /f/ and /θ/. These findings will be discussed and compared to the findings based on intelligibility ratings in Aoyama *et al*. (2008).

3pSC6. Pitch range, intensity, and vocal fry in non-native and native English focus intonation. Alex Hong-lun Yeung, Hyunah Baek, Chikako Takahashi, Joseph Duncan, Sharon Benedett (Dept. of Linguist, Stony Brook Univ., Stony Brook, NY 11794-4376, chikako.takahashi@stonybrook.edu), Jiwon Hwang (Asian & Asian American Studies, Stony Brook Univ., Stony Brook, NY), and Ellen Broselow (Linguist, Stony Brook Univ., Stony Brook, NY)

23 native English speakers (ES) and 25 native Mandarin speakers (MS) participated in a study of the production of contrastive focus prosody in English. The participants completed an interactive game in which they directed experimenters to decorate objects, producing sentences containing contrasting noun phrases in which either the adjective or noun was contrasted (e.g., Andy wants an orange diamond on his towel and a NAVY diamond/orange OVAL on Mindy's towel). Time-normalized average pitch and intensity contours extracted from a subset of the speakers suggest that while both groups distinguish adjective from noun focus, the MSs show a wider pitch range but smaller intensity drop than the ESs, consistent with a previously reported study of contrastive focus production (Takahashi *et al*., 2017). A surprising pattern in the data was that the MSs actually showed a stronger use of pitch cues on the focused noun than the ESs, which may have reflected the fact that many of the ESs exhibited creakiness toward the end of the sentence, restricting their use of pitch to mark focus on nouns. We argue that these divergent patterns reflect a combination of Mandarin L1 influence and innovative vocal fry prosody in native English speakers.

3pSC7. Changes in the phonetics of German language learners over a semester. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com) and Lauren Elliott (Psych., Carthage College, Kenosha, WI)

We compare the acoustic phonetics of German language learners at the beginning and end of a college semester. Participants recorded three utterances of five example sentences in the first and last months of the semester. Results from the early recordings documented that language learners were intermediate between native speakers and naive speakers in at least two different acoustic contexts. The voice onset time (VOT) of the /f/ sound was short in native speakers (more /f/-like) and long in naive speakers (more /v/-like). Similarly, the vowel space traversed during the vowel sound of "viel" had a smaller area in native speakers than in naive speakers. We predict that, on average, language learners will show change in both their VOT and their vowel space, becoming more similar to native speakers, while naive speakers will not show any change. We will use paired t-tests to compare the early and late productions from each speaker. We will also expand the array of acoustic features that we will examine to include additional vowel and consonant sounds.

3pSC8. Interplay of native and non-native vowels in Japanese late learners of English. Chikako Takahashi (Dept. of Linguist, SUNY at Stony Brook, Stony Brook, NY 11794-4376, chikako.takahashi@stonybrook.edu)

This study investigates perception of native (/i/-/e/) and non-native (/i/-/ɪ/) contrasts by Japanese late learners of English (N = 40). We hypothesized that speakers with greater proficiency in L2 might show more effects of L2 learning on L1 perception than speakers with lower L2 proficiency. We found first that self-rated proficiency correlated relatively well with L2 vowel contrast categorization ($r = 0.622$) indicating that late bilinguals can achieve more nativelike categorization of L2 vowels (/i/-/ɪ/) as their overall proficiency improves. Turning to the effect of L2 proficiency on L1 perception, we found that the bilinguals who exhibited more nativelike English /i/-/ɪ/ categorization in their L2 were more likely to have a broader /i/ category when identifying /i/vs./e/ in Japanese (compared to monolingual Japanese controls). We argue that while L2 English learners may be improving in identifying English /i/vs./ɪ/, the non-native /ɪ/ is not fully differentiated from their Japanese /i/ category, possibly providing atypical exemplars that influence the perceptual boundary of /i/ in their L1. The complex interplay of L1 and L2 vowels will be further discussed by reference to production data for the learners and native speaker controls for Japanese and English.

3pSC9. Second langauge production of French mid and high vowels: An articulatory perspective. Madeleine Oakley (Linguist, Georgetown Univ., 3700 O St. NW, Washington, DC 20057, mo643@georgetown.edu)

This study uses Ultrasound Tongue Imaging and acoustic data to investigate the articulatory strategies used by L1 English L2 French learners to produce round vowels. It has been suggested that learners have more difficulty in producing L2 phones that are "similar" to L1 phones than L2 phones that are completely "new" because learners use L1 categories to produce L2 phones (Flege, 1982; Kamiyama and Vaissiere, 2009). However, this claim is based solely on acoustic data. To this end, the present study records learners' articulatory strategies using Ultrasound during production of French round vowels /y, u, ø, o/ compared to English /u, o/. 1 L1 French speaker and 6 L2 French learners were recorded producing wordlists in French and English, using ultrasound, video recordings of lip protrusion, and audio recordings. Results show that learners do not, in fact, use L1 articulatory strategies to produce L2 phones. Additionally, articulatory data show that learners still have difficulty producing target-like tongue positions for new phones, despite having target-like acoustic productions, which may suggest that non-native vowels have an acoustic rather than an articulatory target.

3pSC10. Non-native perception of noise-vocoded speech. Ewa Jacewicz, Robert A. Fox, and Joy Lee (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

Research has established that non-native speakers consistently underperform compared to native speakers in challenging listening environments such as in noisy backgrounds. Furthermore, when presented with variable talkers and regional accents, non-native listeners are unable to benefit from available phonetic details to the same extent as do native listeners [Jacewicz *et al*., *J. Acous. Soc. Am.* **144**, 1864 (2018)]. The current study further inquired into this non-native processing deficit using noise-vocoded speech. Noise-vocoding not only models cochlear implant processing but is also informative regarding perception of degraded speech in normal-hearing listeners. Proficient Korean-English bilinguals were asked to identify talker dialect (Ohio, North Carolina) and talker sex responding to utterances processed through a noise-source vocoder at 4, 8, 12, and 16 channels. A separate task assessed intelligibility of this noise-vocoded speech. Since cues to

talker dialect are found primarily in fine spectral structure, we expected them to perform increasingly worse than native listeners with the decreasing number of channels. Both native American English controls and Korean-English listeners adhered to the predicted pattern across tasks and conditions. The study provides further evidence that the ability to utilize partial spectral information is compromised in non-native speech processing.

3pSC11. Effects of pitch contour and speaking rate on perception of foreign-accented speech. Rebecca F. Davis (Psychol. and Brain Sci., Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, rebecca.davis.1@louisville.edu), Melissa M. Baese-Berk (Linguist, Univ. of Oregon, Eugene, OR), and Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Perception of foreign accent is typically studied using an accentedness rating task. For example, native English listeners rate the degree of accentedness in sentences produced by non-native English speakers. However, in the past studies, it has been unclear that on what criteria participants used to judge accentedness. Here, native English speakers rated the accentedness of Korean-accented English sentences on a scale from 1 (strong accent) to 9 (little to no accent). Participants rated sentences that were unmodified or had one acoustic property removed. In one block, pitch contours of sentences were flattened and set to their mean values. In another block, speaking rates were set to the grand mean of all speaking rates (3.8 syllables/second). This way, changes in accentedness ratings across unmodified and modified sentences were attributable to the acoustic property that was removed. Accentedness ratings were not systematically influenced by manipulations of pitch contours but were influenced by speaking rate manipulations. Increasing the speaking rate (to 3.8 syllables/second) made sentences sound less accented than their unmodified versions; decreasing the speaking rate made sentences sound more accented than their unmodified versions. Results suggest that the speaking rate directly contributes to ratings of foreign accentedness.

3pSC12. Task-evoked pupillary response to completely intelligible accented speech. Drew J. McLaughlin (Psychol. & Brain Sci., Washington Univ. in St. Louis, 1 Brookings Dr., St. Louis, MO 63130, drewjmclaughlin@wustl.edu) and Kristin Van Engen (Psychol. & Brain Sci., Washington Univ. in St. Louis, Saint Louis, MO)

Listening to second language- (L2-) accented speech is often described as an effortful process, even when L2 speakers are highly proficient. This increase in listening effort is likely caused by systematic segmental and suprasegmental deviations from native-speaker norms, which require additional cognitive resources to process (Van Engen and Peelle, 2014). In this view of speech perception, even when an L2 speaker is completely intelligible (i.e., their words can all be correctly identified), perception nonetheless requires increased cognitive load compared to native speech. We used pupillometry (the measure of pupil diameter over time) as a psychophysiological index of cognitive load to address this hypothesis. Task-evoked pupillary response (TEPR) was measured while participants listened to sentences produced by a Mandarin Chinese-accented English speaker and a standard American-accented English speaker. Results from a first experiment showed that TEPR was larger for L2-accented speech than the native speech, indicating greater cognitive load during speech processing. In a second experiment, we controlled for differences in the speech rate between the two speakers. Preliminary evidence from this replication also shows larger TEPR for the L2-accented speech condition. Together, this evidence suggests that processing accented speech—even when completely intelligible—requires additional cognitive resources.

3pSC13. Production and perception of two English vowels by advanced Colombian Spanish learners of English. Jenna Conklin, Olga Dmitrieva (Linguist, Purdue Univ., 610 Purdue Mall, West Lafayette, IN 47907, jconkli@purdue.edu), and Gina M. Pineda Mora (Universidad Nacional de Colombia, Bogota, Colombia)

This study investigates the acquisition of the English vowels [æ] and [ʌ] by Colombian learners of American English. Thirty speakers of Colombian Spanish residing in the United States participated in this study. A multiple-choice identification task demonstrated that participants were proficient at identifying [æ] and [ʌ], though the accuracy rate was somewhat lower than for a number of other English vowels used for comparison. When misidentified, [æ] and [ʌ] were most frequently confused with each other or with the low back vowel [ɑ]. In a production task completed by both bilingual Spanish speakers and monolingual English speakers, learners' [æ] and [ʌ] were distinct from native monolingual English productions. The nature of the acoustic differences between native speakers' vowels and learners' vowels suggested that Colombian speakers' English renditions of [æ] and [ʌ] were affected by their Spanish [a]. Further analysis revealed that learners' [æ] was not statistically different from their own production of Spanish [a]. These results are compatible with the assumption emerging from previous research that speakers of Spanish acquiring English assimilate, to a varying degree, English low and low-mid vowels to Spanish [a]—a single vowel found in this area of the vowel space in Spanish.

3pSC14. Changes to perceptual assimilation following training. Julia Kharlamenko, Heather M. Kabakoff (Communicative Sci. and Disord., New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, jk5096@nyu.edu), Erika Levy (Commun. Sci. and Disord., Columbia Univ., New York, NY), and Susannah V. Levi (Communicative Sci. and Disord., New York Univ., New York, NY)

Adult speakers of American-English have difficulty perceiving front-back rounding contrasts in vowels, such as French /œ/-/o/. These difficulties stem in part from how learners map sounds in a second language onto native-language sound categories. We used a perceptual-assimilation task to test how perceptual training changes listeners' mapping of nonnative speech sounds. One listener group was trained with a bimodal distribution of stimuli drawn from an /œ/-/o/ acoustic continuum, a second was trained with a unimodal distribution, and a third (control) group completed no training. The training incorporated active learning with feedback and lexical support. In the assimilation task, listeners heard French vowels and were asked to select which of eight English /hVd/ words best matched the French stimulus. Results revealed general perceptual training effects and differences in both trained groups. For /o/, both trained groups differed significantly from untrained controls. However, for /œ/, listeners in the unimodal group did not differ from the untrained controls, but the bimodal group differed from the untrained controls. Thus, exposure to a unimodal distribution can alter perception, but a bimodal distribution may have stronger effects on perception. Bimodal distribution may be better for facilitating assimilation of nonnative sounds to native-like categories.

3pSC15. A comparison between native and non-native speech for automatic speech recognition. Seongjin Park and John Culnan (Dept. of Linguist, University of Arizona, Box 210025, Tucson, AZ 85721, seongjinpark@email.arizona.edu)

This study investigates differences in sentence and story production between native and non-native speakers of English for use with a system of Automatic Speech Recognition (ASR). Previous studies have shown that production errors by non-native speakers of English include misproduced segments (Flege, 1995), longer pause duration (Anderson-Hsieh and Venkatagiri, 1994), abnormal pause location within clauses (Kang, 2010), and non-reduction of function words (Jang, 2009). The present study uses phonemically balanced sentences from TIMIT (Garofolo *et al.*, 1993) and a story to provide an additional comparison of the differences in production by native and non-native speakers of English. Consistent with previous research, preliminary results suggest that non-native speakers of English fail to produce flaps and reduced vowels, insert or delete segments, engage in more self-correction, and place pauses in different locations from native speakers. Non-native English speakers furthermore produce different patterns of intonation from native speakers and produce errors indicative of transfer from their L1 phonology, such as coda deletion and vowel epenthesis. Native speaker productions also contained errors, the majority of which were content-related. These results indicate that difficulties posed by English ASR systems in recognizing non-native speech are due largely to the heterogeneity of non-native production.

3pSC16. The development of perception of English tense-lax vowel contrast by monolingual English-speaking and bilingual Mandarin-English children. Lan Yu (Indiana Univ. Bloomington, 1020 E. Kirkwood Ave., Ballantine 821, Bloomington, IN 47405, ly29@indiana.edu)

This study explored the perceptual weighting utilized by children and adults for the perception of the English tense-lax vowel contrast in both the first language (L1) and second language (L2). Both the desensitization hypothesis (Bohn, 2000), which posits a universal bias toward duration cues on non-native vowel contrasts, and developmental perceptual weighting shift hypothesis in L1 (Nittrouer *et al*., 1993), which suggests a shift from dynamic to static cues developmentally, were tested in this study. Listeners were 4 English monolingual children (EC) with a mean age of 6:9, 4 Mandarin-English bilingual children (MEC) with a mean age of 8:4, 4 English-speaking adults (EA), and 4 Mandarin-English bilingual adults (MEA). In an identification experiment, listeners identified stimuli from *beat*/*bit* and *sheep*/*ship* continua differing in six acoustically equal duration and spectral steps. EC, MEC, and EA rely on spectral differences exclusively, yet MEA utilize an additional cue, duration, to identify this contrast. Findings suggest that desensitization needs to be interpreted with caution since other factors might greatly shift the perceptual weighting such as the age of L2 acquisition and the amount of L2 exposure. As for a developmental perceptual weighting shift, it only applies to the perception of consonants not vowels.

3pSC17. Non-native perception and first language experience: The case of English final stop perception by Saudi Arabic listeners. Sarah Alamri (GMU, 4400 University Dr., Fairfax, VA 22030, salamri4@gmu.edu)

This study investigated how listeners' native language [Saudi Arabic (SA)] could affect the perception of non-native (English) sound patterns. SA stopsare generally released, while English stops show free variation: released or unreleased. In a phoneme detection task, SA listeners were asked to detect English stops in coda position, and their accuracy and reaction time (RT) were measured. The influence of three factors of the final coda (i.e., release, voicing, and place of articulation) on the accuracy and RT was statistically analyzed using mixed effect models. The results indicated a significant interaction among the three factors on the listeners' accuracy but not on the RT. The results revealed that listeners relied mostly on release bursts as a phonetic cue to accurately detect the target phoneme. The significant interaction between release and voicing suggested that the unreleased voiced stops were easier to detect than the voiceless counterparts. Despite the difference in the accuracy, the listeners' RT was not significantly influenced by release, voicing, or place of articulation. The results suggested that L1 experience effects non-native perception if the target phoneme does not have enough acoustic information. Listeners can accurately detect unfamiliar sound patterns when the signal is acoustically rich.

3pSC18. Formant discrimination of speech and non-speech sounds in temporally modulated noise: Effect of language experience. Mingshuang Li, Can Xu, Chang Liu (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX 78712, limingshuang@utexas.edu), and Sha Tao (State Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Beijing, China)

Recent studies showed that the residency experience of 1–3 year in English-speaking countries could improve temporal dip listening against energetic masking of noise (i.e., masking release), resulting in better performances in identification of vowel, consonant, and sentence in temporally fluctuating noise. The current study aimed to explore whether this advantage would be extended to speech and non-speech perception discrimination. Three groups of listeners were recruited: English-native listeners (EN), Chinese-native listeners in US with long US residency experience (i.e., 2.5–4 years; CNUL), and Chinese-native listeners in the US with short US residency experience (i.e., less than half a year; CNUS). Thresholds of spectral discrimination of speech and non-speech stimuli were measured. EN listeners are expected to have the lowest thresholds and highest masking release in vowel formant discrimination, followed by CNUL listeners, with highest thresholds and lowest masking release for CNUS listeners. Furthermore, the current study will also examine whether the effect of language experience on masking release would be presented for non-speech spectral discrimination in noise. [Work supported by the China National Natural Science Foundation (31628009).]

3pSC19. Acoustic analysis of nasal and lateral consonants: The merger in Eastern Min. Ruoqian Cheng and Allard Jongman (Dept. of Linguist, Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045, rqcheng@ku.edu)

The contrast between word-initial [n] and [l] is disappearing in many Chinese languages, including Eastern Min. Before investigating the status of the merger, we need to identify the acoustic cues that distinguish [n] from [l]. In English and Mandarin, languages with the [n] and [l] contrast, we examined: (1) Duration: consonant duration and consonant-vowel transition duration; (2) Formant frequencies: F1, F2 and F3 at the midpoint of the consonant; (3) Formant intensities: I1, I2, and I3 at the midpoint of the consonant; and (4) Relative amplitude (the amplitude difference between the consonant and the following vowel). Preliminary results show that [n] is significantly longer than [l] in Mandarin and English but not in Eastern Min. F2 of [n] is higher than F2 of [l] in English and Mandarin, whereas the direction is reversed in Eastern Min. Finally, the difference in relative amplitude between [n] and [l] is greater in English and Mandarin than in Eastern Min. Together, these results suggest a merger in progress in Eastern Min . Moreover, older Eastern Min speakers showed the merger to a greater degree than younger speakers, presumably because the older speakers use Mandarin less frequently.

Plenary Session and Awards Ceremony

Lily M. Wang,
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New Fellows

Megan S. Ballard – For contributions to shallow water propagation and geoacoustic inversion

Woojae Seong – For contributions to geoacoustic inversion and ocean signal processing

Lori J. Leibold – For contributions to our understanding of auditory development

Robert W. Pyle, Jr. – For contributions to the understanding of the acoustics of brass musical instruments

Rajka Smiljanic – For contributions to cross-language speech acoustics and perception

Edward J. Walsh – For contributions to auditory physiology, animal bioacoustics, and public policy

Introduction of Award Recipients and Presentation of Awards

Rossing Prize in Acoustics Education to Stanley A. Chin-Bing

William and Christine Hartmann Prize in Auditory Neuroscience to Glenis R. Long

Distinguished Service Citation to David Feit

R. Bruce Lindsay Award to Adam Maxwell

Silver Medal in Engineering Acoustics to Thomas B. Gabrielson

Helmholtz-Rayleigh Interdisciplinary Silver Medal to Barbara G. Shinn-Cunningham

Gold Medal to William J. Cavanaugh

Vice President's Gavel to Scott D. Sommerfeldt

President's Tuning Fork to Lily M. Wang

Session 3eED

Education in Acoustics and Women in Acoustics: Listen Up and Get Involved

L. Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Ste. 300, Melville, NY 11747

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

This workshop for Louisville area Girl Scouts (age 12–17) consists of hands-on tutorials, interactive demonstrations, and discussion about careers in acoustics. The primary goals of this workshop are to expose girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please e-mail Keeta Jones (kjones@acousticalsociety.org) if you have time to help with either guiding the girls to the event and helping them get started (5:00 p.m. to 6:00 p.m.) or exploring principles and applications of acoustics with small groups of girls (5:00 p.m. to 7:30 p.m.). We will provide many demonstrations but feel free to contact us if you would like to bring your own.

Chair's Introduction—5:00

WEDNESDAY EVENING, 15 MAY 2019 7:30 P.M. TO 9:00 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics which will meet starting at 4:30 p.m.

These are working, collegial meetings. Much of the work of the Society is accomplished by actions that originate and are taken in these meetings, including proposals for special sessions, workshops, and technical initiatives. All meeting participants are cordially invited to attend these meetings and to participate actively in the discussion.

Committees meeting on Tuesday are as follows:

Engineering Acoustics (4:30 p.m.)	McCreary
Signal Processing in Acoustics (4:30 p.m.)	Beckham
Acoustical Oceanography	McCreary
Animal Bioacoustics	Clements
Architectural Acoustics	French
Musical Acoustics	Breathitt
Physical Acoustics	Jones
Psychological and Physiological Acoustics	Carroll Ford
Structural Acoustics and Vibration	Stopher

Committees meeting on Wednesday

Biomedical Acoustics	Nunn

Committees meeting on Thursday

Computational Acoustics (4:30 p.m.)	Clements
Noise	Segell
Speech Communication	Carroll Ford
Underwater Acoustics	McCreary

Distinguished Service Citation

David Feit

2019

The Distinguished Service Citation is awarded to a present or former member of the Society in recognition of outstanding service to the Society.

PREVIOUS RECIPIENTS

Laurence Batchelder	1972	William J. Cavanaugh	1994
Robert W. Young	1973	John C. Burgess	1996
Betty H. Goodfriend	1973	Alice H. Suter	1997
Gerald J. Franz	1974	Elaine Moran	1999
Robert T. Beyer	1978	John V. Bouyoucos	2000
Henning E. von Gierke	1978	F. Avril Brenig	2000
R. Bruce Lindsay	1981	Thomas D. Rossing	2006
William S. Cramer	1984	Charles E. Schmid	2008
Stanley L. Ehrlich	1986	Uwe J. Hansen	2011
Samuel F. Lybarger	1986	Richard Stern	2011
Frederick E. White	1987	Allan D. Pierce	2015
Daniel W. Martin	1989	Paul D. Schomer	2015
Murray Strasberg	1990	Susan B. Blaeser	2016

ENCOMIUM FOR DAVID FEIT

. . . for service and contributions to the Acoustical Society of America, especially as Treasurer

LOUISVILLE, KENTUCKY • 15 MAY 2019

David Feit has a long history of service to national defense, to the study of acoustics and to the Acoustical Society of America (ASA). David is known in the acoustics community for his mastery of acoustics, and his wonderful writing and teaching skills.

David began his studies in engineering at Columbia University, receiving four degrees and culminating in an Engineering Science Doctorate in 1964. As a graduate student, he worked at the Stevens Institute of Technology Experimental Towing Tank, which began his long career in working with ships. After graduation, David worked at Cambridge Acoustical Associates (CAA), Inc., and then joined in 1973 the Carderock Division as the Head of the Vibrations and Acoustics Technology Division of the Ship Acoustics Department. He also served as the Liaison Scientist for Acoustics and Mechanics at the Office of Naval Research European Office from 1988 to 1990. David's service to the Navy was so exemplary that he received a letter of appreciation from the Commander of the Naval Sea Systems Command and the Navy's Meritorious Civilian Service Award in 1982.

Not only did David serve the Navy as a valued researcher, he has also served the acoustics academic community as an Adjunct Professor at Catholic University, George Washington University, Naval Academy, and University of Maryland for varying periods. David also lectured at the Massachusetts Institute of Technology (MIT) on the "Fundamentals of Ship Acoustics" in the summers of 1982 to 1986 and from 1993 to 1998. David is an author of 31 papers, 17 of which were published in the *Journal of the Acoustical Society of America*. He was recognized for his considerable achievements with the Trent-Crede Medal from the ASA in 1999, and the Per Bruel Gold Medal from the American Society of Mechanical Engineers in 2003.

David's most lasting contribution to the acoustics community is his book, *Sound, Structures and Their Interaction*, which was coauthored with Miguel C. Junger. This book is a superbly well-written, seminal work and has had an immeasurable impact on our community. It can be a challenge to find a superfluous sentence in either this book or in any of David's numerous technical papers and presentations, which are all clear, concise, and focused. (David also has a reputation for using equations on napkins as common form of communication sometimes to the dismay of family and restaurant waiters.) These qualities spill over to David's technical discussions with colleagues and, when combined with his modest and unassuming manner, earned him a (closet) sobriquet. Rather than by his initials DF, David was known as EF, mimicking the commercial tag line popular in the 1970's, "When EF Hutton talks, people listen".

Lastly, David has been an exemplar for service to our society. He joined the Society in 1965 and was elected a Fellow in 1975. David began participating in ASA meetings as a presenter and session organizer in the 1960s, and served as Technical Program Chair for the spring 1995 meeting held in Washington, DC. David has been a member of the Technical Committee on Structural Acoustics and Vibration (TCSA) since 1965, serving as its chair from 1992 to 1998. As the TCSA chair he was also a member of the Technical Council.

David has served as associate editor of the *Journal of the Acoustical Society of America* since 2005. In 1993, he and Miguel Junger donated the copyright of *Sound, Structures and Their Interaction* to the ASA so it could be included in its classic works series.

However, David's most important and significant role in the Society has been as the ASA treasurer since 2000. David managed ASA's 5 million dollar annual budget and provided guidance to the Executive Council on the Society's financial affairs. As treasurer, he served as *ex officio* on seven committees related to ASA's financial endeavors. He has tirelessly worked with other officers, committee chairs, staff members, and the ASA accountants and auditors to insure that the ASA's finances are handled efficiently and accurately. During my

term (M.I.) as president, I observed, firsthand, David's commitment to the financial health of the ASA.

David's reputation is worldwide. Whether from his Columbia University days, his stints at CAA and the David Taylor Research Center, or his European assignment, David has colleagues, acquaintances and friends everywhere.

David's long standing devotion and service to the acoustics community and the ASA, make this award very well deserved, and it could not be going to a finer person. And, as we all know, "When DF Feit talks, people listen".

MARCIA ISAKSON
JOEL GARRELICK
SUSAN FOX
JAMES LYNCH

R. BRUCE LINDSAY AWARD

Adam Maxwell

2019

The R. Bruce Lindsay Award (formerly the Biennial Award) is presented in the Spring to a member of the Society who is under 35 years of age on 1 January of the year of the Award and who, during a period of two or more years immediately preceding the award, has been active in the affairs of the Society and has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics, or both. The award was presented biennially until 1986. It is now an annual award.

PREVIOUS RECIPIENTS

Richard H. Bolt	1942	Joseph M. Cuschieri	1991
Leo L. Beranek	1944	Anthony A. Atchley	1992
Vincent Salmon	1946	Michael D. Collins	1993
Isadore Rudnick	1948	Robert P. Carlyon	1994
J. C. R. Licklider	1950	Beverly A. Wright	1995
Osman K. Mawardi	1952	Victor W. Sparrow	1996
Uno Ingard	1954	D. Keith Wilson	1997
Ernest Yeager	1956	Robert L. Clark	1998
Ira J. Hirsh	1956	Paul E. Barbone	1999
Bruce P. Bogert	1958	Robin O. Cleveland	2000
Ira Dyer	1960	Andrew J. Oxenham	2001
Alan Powell	1962	James J. Finneran	2002
Tony F. W. Embleton	1964	Thomas J. Royston	2002
David M. Green	1966	Dani Byrd	2003
Emmanuel P. Papadakis	1968	Michael R. Bailey	2004
Logan E. Hargrove	1970	Lily M. Wang	2005
Robert D. Finch	1972	Purnima Ratilal	2006
Lawrence R. Rabiner	1974	Dorian S. Houser	2007
Robert E. Apfel	1976	Tyrone M. Porter	2008
Henry E. Bass	1978	Kelly J. Benoit-Bird	2009
Peter H. Rogers	1980	Kent L. Gee	2010
Ralph N. Baer	1982	Karim G. Sabra	2011
Peter N. Mikhalevsky	1984	Constantin-C. Coussios	2012
William E. Cooper	1986	Eleanor P. J. Stride	2013
Ilene J. Busch-Vishniac	1987	Matthew J. Goupell	2014
Gilles A. Daigle	1988	Matthew W. Urban	2015
Mark F. Hamilton	1989	Megan S. Ballard	2016
Thomas J. Hofler	1990	Bradley E. Treeby	2017
Yves H. Berthelot	1991	Yun Jing	2018

CITATION FOR ADAM D. MAXWELL

. . . for contributions to the understanding and application of therapeutic ultrasound

LOUISVILLE, KENTUCKY • 13 MAY 2019

Adam Maxwell grew up in the Seattle area in a family entrenched in the sciences. His grandfather was an aerospace engineer for Boeing and his father a development engineer at Philips/ATL. Adam followed suit and enrolled in electrical engineering at the University of Washington with a focus on circuits, devices and technology. His introduction to research came during his sophomore year when a family friend with connections to the Applied Physics Laboratory encountered Adam ringing-up groceries as his part-time job. She thought the kid's talents might be put to better use. Bob Odom, then at the APL, steered Adam to Mike Bailey's lithotripsy research lab. Adam was impressive from the very start and it wasn't long before he was building custom equipment and repairing the lithotripter high-voltage supply. It is noteworthy that as an undergraduate Adam led the way in developing an inexpensive PVDF membrane hydrophone that proved to be an accurate and reliable alternative to the fiber-optic probe hydrophone used to measure lithotripter shock waves. Adam built dozens of these hydrophones, designed and made preamplifiers for them, tested sensitivity and modeled their frequency response. His design was subsequently licensed by U-Washington to a Seattle company that successfully marketed the device. Before he left for graduate school Adam also played a key role in a seminal research study demonstrating the action of shear waves as a mechanism for stone fracture in shockwave lithotripsy.

At the University of Michigan, Adam was awarded an NSF fellowship to pursue his PhD in Biomedical Engineering under the supervision of Zhen Xu. Here, Adam found his niche in research on acoustic cavitation. In this work Adam made key contributions to the understanding of bubble cloud formation in histotripsy, the mechanical disruption of tissue by short pulses of high amplitude ultrasound. His work provided the first evidence that peak positive pressure contributes to the cavitation cloud responsible for histotripsy erosion and that the high frequency content of the shock wave was significant in reflecting and inverting the pressure from the initial bubble. Adam discovered he could create histotripsy with acoustic pulses having predominant negative pressures and leveraged this observation to pursue an intrinsic cavitation threshold for fluids and tissue-like materials. This led to the demonstration that histotripsy erosion depended on tissue type and that certain cancers could be eroded while leaving otherwise healthy tissue and blood vessels intact. Adam also developed histotripsy techniques to disrupt blood clots and discovered that histotripsy could be used to make dense collagenous connective tissue more compliant, a finding he would build on years later to demonstrate proof of principle in softening fibrous scars to recover tissue flexibility in Peyronie's disease. Working with Tim Hall, one of the co-inventors of histotripsy, Adam helped develop 3D printable transducers that are now ubiquitous in histotripsy and lithotripsy research. His deep understanding of histotripsy and his expertise with the instrumentation to produce it made him a valuable consultant to HistoSonics, a company spun out by U-Michigan to commercialize histotripsy. In addition, Adam investigated small-scale interactions between solitary bubbles and cells leading to the generation of acoustic streaming to trap small particles. Overall, Adam's studies combined aspects of shock formation, radiation force and acoustic streaming to the understanding of how ultrasound at extreme pressure can generate and grow a cavitation cloud and how the cloud erodes tissue and tissue-like materials.

At the completion of his graduate studies, Adam was attracted back to U-Washington by the opportunity to contribute to a rapidly growing partnership between the APL, the Center of Industrial and Medical Ultrasound and the Department of Urology. He turned his attention to applying cavitation to the treatment of stone disease. Adam built upon his mechanistic understanding of stone breakage to invent and develop burst wave lithotripsy (BWL), a direct competitor to conventional shock wave lithotripsy. Within 5 years

of Adam's invention, BWL is the key technology of a spin-off company (SonoMotion) that is already testing BWL in human patients and is working toward FDA regulatory clearance to market a device. This technology has also been integrated into the combined diagnostic and therapeutic ultrasound system NASA is developing for space exploration.

Adam is a builder intent on translating new technologies into practice. He chooses challenging, significant problems and finds answers through well-designed and carefully conducted research. His aim has been to develop and refine innovative ultrasound devices that will improve treatment outcomes for patients over a spectrum of clinical indications and applications. Adam has co-authored over 35 issued or pending patents covering many of his discoveries. In his young career he has already played an essential role in establishing foundational technologies for three biotech and medical device companies, HistoSonics for the non-invasive, non-thermal ablation of cancerous tumors; Matchstick Technologies for cavitation-based fragmentation of DNA in multi-well assay plates; and SonoMotion for the non-invasive fragmentation of urinary stones using BWL. Adam is also developing an acoustic tractor beam with which he has successfully trapped, lifted and carried urinary stones through the pig bladder. Such a technology holds unique potential for use in clearing stones from the kidney, non-invasively capturing and steering them through the complex three-dimensional path of the urinary space.

Adam has long been active in service to the ASA. He is a highly engaged member of the Biomedical Acoustics Technical Committee, has chaired several ASA special sessions and has authored two *Acoustics Today* articles, one on lithotripsy and one on histrotripsy. He reviews manuscripts regularly for a dozen journals, including enough JASA papers to be on the editors' thank you list. Those who know Adam appreciate not only his capabilities as a researcher and critical thinker but view him fondly and with great respect for his willingness to work hard to help others succeed. Adam is exceptionally creative, but he is also selfless, genuine and generous.

On behalf of those who have had the privilege of working with Adam we welcome this opportunity to celebrate his many accomplishments. Adam D. Maxwell is in all respects a colleague highly deserving of the 2019 R. Bruce Lindsay Award.

MICHAEL R. BAILEY
J. BRIAN FOWLKES
JAMES A McATEER

Silver Medal in Engineering Acoustics

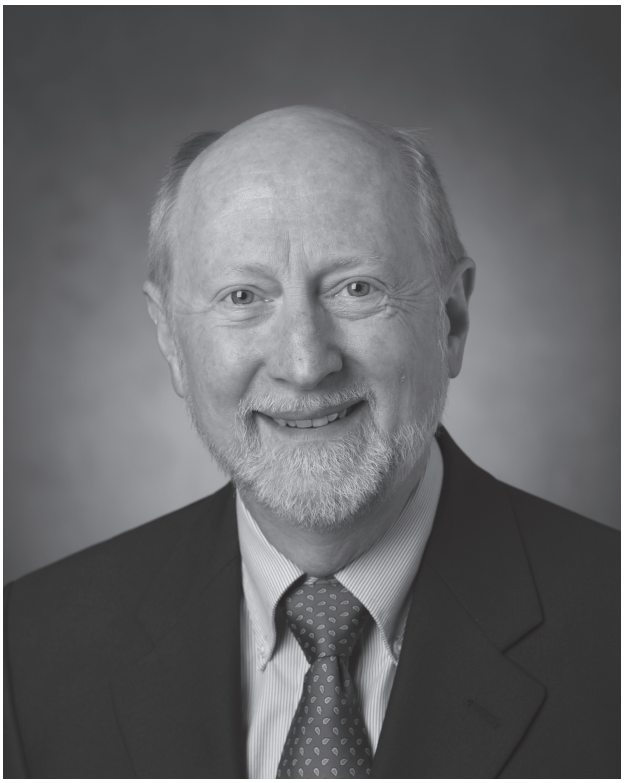
Thomas B. Gabrielson

2019

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Harry F. Olson	1974	James E. West	1995
Hugh S. Knowles	1976	Richard H. Lyon	1998
Benjamin Bauer	1978	Ilene J. Busch-Vishniac	2001
Per Vilhelm Briiel	1982	John V. Bouyoucos	2004
Vincent Salmon	1984	Allan J. Zuckerwar	2007
Albert G. Bodine	1986	Gary W. Elko	2011
Joshua E. Greenspon	1989	John L. Butler	2015
Alan Powell	1992		



CITATION FOR THOMAS B. GABRIELSON

. . . for contributions to the understanding of novel transducers and their intrinsic limitations imposed by thermal and quantum physics

LOUISVILLE, KENTUCKY • 11 MAY 2019

More than most members in the Acoustical Society of America (ASA), Tom Gabrielson is perceived to be someone who works in the same field as yours. If you study shallow water propagation and use sonobuoys, you think that must be Tom's specialty. If your work involves micromachined sensors, you think that's his specialty. Same story for atmospheric propagation and infrasound. For the correct perspective, you only need to see Tom as a fully dedicated scientist and experimentalist. Whether the project is sponsored research or just a source for his own personal amusement, Tom brings the same innovative talent and attention to detail. One independent project of this type was a seismic sensing system installed at his Pennsylvania home that was capable of detecting waves crashing on the New Jersey shore some seventy miles away.

Tom received a B.S.E.E. from the New Jersey Institute of Technology and M.S. and Ph.D. degrees from Pennsylvania State University. He is also a Licensed Professional Engineer in Pennsylvania and is Board Certified by the Institute of Noise Control Engineering. Tom began his professional career in 1974 at the Naval Air Development Center (NADC) in Warminster, PA. While completing his Ph.D., he spent a year as a Visiting Professor at the Naval Postgraduate School in Monterey, CA, and then returned to NADC. At NADC, he developed several models for the prediction of acoustic propagation underwater, and led the Naval Air Warfare Center Harsh Environments Program to measure acoustic propagation under environmentally complex conditions. As a flight-rated project specialist, he collected acoustic data around the globe in P-3 aircraft. While at NADC, Tom was the recipient of both their Scientific Achievement Award and Outstanding Independent Research Project Award. He was also elected a Fellow of ASA in 1995. After NADC closed in 1996, Penn State was fortunate to hire Tom into the Graduate Program in Acoustics and the Applied Research Laboratory where he currently serves as Professor of Acoustics and Senior Scientist.

Tom's scientific interests have always been eclectic, ranging from astronomy and geosciences to many acoustical sub-disciplines. Constant throughout has been Tom's enjoyment of the design, fabrication, and testing (i.e., engineering) of complete instrumentation systems, ranging from the raw sensor and its housing, through the signal-conditioning electronics, to the calibration and subsequent signal-processing at the end of any transduction chain. In addition to this lab-focused effort, he also has an addiction to field experiments that exploit his custom transducer systems. Tom's field experiments have included measurement of underwater sound propagation in shallow water, military jet noise studies, and acoustic measurement of solid transport by mountain streams. His field experiment that received the most public attention was his work with a team from Penn State to locate survivors after the World Trade Center collapse using remotely deployable microphones and accelerometers connected to signal-conditioning electronics and recording devices built into a backpack. More recently, his measurements of the noise produced by natural-gas compressor stations was featured on National Public Radio.

Tom is best known in the Engineering Acoustics community for his contributions to the development, calibration, test, and analysis of both conventional and unconventional transducers, as well as conventional transducers used in unconventional contexts, such as the geophones used as underwater acoustic particle velocity detectors. He has investigated thermoacoustic engines as underwater sound sources, electron tunneling junctions as miniature accelerometers, and accelerometer-based acoustic intensity sensors.

Tom has been the author or coauthor of papers in the *IEEE Transactions on Electronic Devices*, ASME *Journal of Vibration and Acoustics, AIAA Journal, US Navy Journal of*

Underwater Acoustics, and the *Journal of the Acoustical Society of America*. He holds four patents and has been an invited speaker at numerous meetings and workshops sponsored by the Office of Naval Research, American Society of Mechanical Engineers, Institute of Noise Control Engineering, American Geophysical Union, American Vacuum Society, and ASA.

Perhaps his broadest impact on our contemporary understanding of transduction is based on his careful analysis of the fundamental limitation imposed by physical noise sources to the signal-to-noise ratio of micro-electro-mechanical sensors (MEMSs) that are fabricated using the technology developed for large-scale integrated circuits. A measure of that influence is the number of citations (417), patent references (35), and electronic downloads (2,694) of his now classic paper entitled "Mechanical-thermal noise in micro-machined acoustic and vibration sensors" (*IEEE Transactons on Electronic Devices* 40(5), May 1993).

Over the last several years, Tom's focus has shifted to the measurement and analysis of atmospheric infrasound. Again, it is the quality of the novel transducers, suppression of wind noise, and such transducers' absolute calibration in the lab and *in situ* calibrations in the field that has become the latest beneficiary of his unique combination of attention to the transduction mechanism and the use of innovative acoustical pre-filtering, signal-conditioning electronics, and subsequent signal processing that has become the hallmark of his unified approach. Tom has also explored the scientific and historical context of those measurements, having written articles on the 1883 eruption of Krakatoa and the history of atmospheric acoustic propagation for *Acoustics Today*.

Although ASA awards a separate prize for acoustics education, this encomium would not be complete if it did not recognize Tom's contributions to Engineering Acoustics through the two graduate courses he has offered each year in Penn State's Graduate Program in Acoustics, and many other short courses. Anyone who has ever had the pleasure of attending one of Tom's presentations can appreciate the unique insight, clarity, originality, and wry humor he brings to such talks. His students' enthusiasm for his lectures, and particularly his lecture demonstrations and homework assignments that involve field experiments, is legendary. He is the only faculty member we know, who has ever received a standing ovation at the end of his course and has received such appreciation in classes over the span now of 30 years.

Tom has been an invaluable asset to the community of Engineering Acoustics, and to the entire Acoustical Society, for nearly forty years. It is fitting that his name should be added to the list of Silver Medal recipients "*for contributions to the understanding of novel transducers and their intrinsic limitations imposed by thermal and quantum physics*."

STEVEN L. GARRETT
DAVID L. GARDNER
JAMES F. McEARCHERN
ARTHUR W. HORBACH
TIMOTHY M. MARSTON

Helmholtz-Rayleigh Interdisciplinary Silver Medal
in
Psychological and Physiological Acoustics, Speech Communication, and Architectural Acoustics

Barbara G. Shinn-Cunningham
2019

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler	1997	James V. Candy	2008
David E. Weston	1998	Ronald A. Roy	2010
Jens P. Blauert	1999	James E. Barger	2011
Lawrence A. Crum	2000	Timothy J. Leighton	2013
William M. Hartmann	2001	Mark F. Hamilton	2014
Arthur B. Baggeroer	2002	Henry Cox	2015
David Lubman	2004	Armen Sarvazyan	2016
Gilles A. Daigle	2005	Blake S. Wilson	2017
Mathias Fink	2006	Kenneth S. Suslick	2018
Edwin L. Carstensen	2007		

Interdisciplinary Silver Medal

Eugen J. Skudrzyk	1983
Wesley L. Nyborg	1990
W. Dixon Ward	1991
Victor C. Anderson	1992
Steven L. Garrett	1993

CITATION FOR BARBARA G. SHINN-CUNNINGHAM

. . . for contributions to understanding the perceptual, cognitive, and neural bases of speech perception in complex acoustic environments

LOUISVILLE, KENTUCKY • 13 MAY 2019

Barbara Shinn-Cunningham attended Brown University as an undergraduate, receiving her B.Sc. in Electrical Engineering in 1986. She then attended the Massachusetts Institute of Technology (MIT) for her M.S. and Ph.D. degrees in Electrical and Computer Engineering. At MIT, Barb worked with Nat Durlach, Pat Zurek, and others in the Research Lab of Electronics, studying spatial hearing. Her work focused on temporal and spatial interactions, including the precedence effect and adaptation to changes in received sounds due to factors such as reverberation, head orientation, and visual inputs. She used both psychoacoustic experiments and mathematical models to understand listeners' capabilities, including adaptation to displaced sources and receivers (heads). Barb received her Ph.D. in 1994 and became a mother shortly after graduation, having the first of her two boys, Nick and Will, with her husband Rob Cunningham. Barb pursued her family life along with her wide interests while she worked part-time as a post-doc, including time at the Sensimetrics Corporation in the Boston area. She returned to academia by joining Boston University (BU) in 1997 with a primary appointment in Cognitive and Neural Systems and a joint appointment in Biomedical Engineering.

Professor Shinn-Cunningham thrived at BU, where she became full Professor of Biomedical Engineering in 2008. In 2011, she established the Center for Computational Neuroscience and Neural Technology as its Founding Director. In 2016, she established the Center for Research in Sensory Communications and Neural Technology, again as Founding Director. Her work with these research centers brought together researchers and graduate students from several academic departments. These centers stimulated interactions among a broad cross section of researchers at BU and beyond. In 2018, Professor Shinn-Cunningham left BU to join Carnegie-Mellon University (CMU) as Director of its new Neuroscience Institute and was appointed Professor of Biomedical Engineering, Psychology, and Electrical and Computer Engineering.

Professor Shinn-Cunningham's research interests broadened over her career from early interest in electrical instrumentation, to hearing psychophysics, to the neurophysiological basis of hearing abilities, and to the development of an integrated understanding of the interactions of all of these aspects with the natural world of communication. Her interests have increasingly been drawn to speech communication, and to the processing of those acoustic signals that are of central importance to human culture. Professor Shinn-Cunningham uses a broad set of empirical and mathematical tools to address speech and hearing in complex situations, including listening tasks that involve multiple speech sources in complex environments with both auditory and visual inputs. She studied the dynamics of onsets and offsets of sources, effects of reverberation, and the effects of cross-sensory and *a priori* knowledge. Her listening tasks involve speech understanding and incorporate multiple aspects of perceptions such as attentional control and visual cues. She also broadened her work with bilateral stimulation in complex environments to include electroencephalography and brainstem potentials. She has extended her work and interests to include a broad cross-section of experimental subjects: normal-hearing listeners, listeners with multiple types of hearing losses (including "hidden hearing losses"), and listeners with more central impairments including challenges like autism and even traumatic brain injury from blast exposure. From a conceptual perspective, this work involves basic mechanisms of speech and hearing, including basic neural coding of waveforms, as well as complex processing and hypothesis testing that allows listeners to separate independent sources, even in environments with reflections and coupled visual inputs.

In her integrative approach to understanding and separating out these multiple effects, Professor Shinn-Cunningham has a strong record of ongoing grant support from multiple

agencies and organizations, already including the National Institute of Deafness and Other Communication Disorders, the National Science Foundation, the Office of Naval Research, and private foundations. Her publications are numerous, over 114 articles in peer-reviewed journals, 48 papers in conference proceedings, and 10 book chapters. Her citations approach ten thousand.

Professor Shinn-Cunningham has actively served the profession through participation in several organizations. Her service to the ASA began with election to the Technical Committee on Psychological and Physiological Acoustics in 2000 and continued with multiple administrative committees. She was elected to its Executive Council (2010-2013), and subsequently as Vice President (2013-2016). She currently serves on the ASA Finance Committee.

Along with her array of intellectual and technical skills, Professor Shinn-Cunnigham cares deeply about her students and their development as scientists. When she was named recipient of the ASA Student Council Mentor Award, the nominators were effusive in their praise of her enthusiasm and excitement for what she, and they, were able to accomplish both professionally and personally. Working with her, one quickly realizes how much she truly loves what she does and how infectious that zest can be. Her enthusiasm for new scientific directions, and her ability to think about questions large and small has helped guide some of the best of the next generation of scientists.

One of the most important aspects of her work, which is very much in the spirit of the Helmholtz-Rayleigh Award, is Professor Shinn-Cunningham's efforts in organizing sessions and conferences that allow discussions by people from a diverse group of research backgrounds and interests. She not only works across a span of research areas, but she also stimulates sharing, comparisons, and collaborations among researchers in different areas. This is reflected in the many special sessions that she has organized and often chaired, representing many environments and organizations (including ASA). This is a natural outgrowth of her grants and contributions across multiple fields, including psychophysical measurements and modeling, neurophysiological recordings and modeling, speech communication characterization both experimentally and conceptually, and the behavioral impact of architectural acoustics. Overall, Professor Shinn-Cunningham is a consummate collaborator. Some of these collaborations she no doubt initiated herself, but others are instances in which she was invited to participate, because of her wide experience and insight. She is a wonderful colleague for many, with diverse ideas, connections, and passions across fields.

In her personal activities, Barb shows a remarkable diversity. She plays the oboe and the English horn and was recently the featured soloist with the Concord Orchestra. She is also an avid fencer and enjoys traveling and the out of doors with her family. Somehow she fits all this into a life that is incredibly productive and that includes many people. She is a wonderful colleague for many, with diverse ideas, connections, and passions across fields.

It is entirely appropriate that the Acoustical Society of America presents the Helmholtz-Rayleigh Interdisciplinary Silver Medal to Barbara G. Shinn-Cunningham–a remarkable scientist who has had enormous influence within the psychoacoustic, physiological, and speech-communication communities as well as the wider neuroscience field.

H. STEVEN COLBURN
PEGGY B. NELSON
WILLIAM M. HARTMANN

Gold Medal

William J. Cavanaugh

2019

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

PREVIOUS RECIPIENTS

Wallace Waterfall	1954
Floyd A. Firestone	1955
Harvey Fletcher	1957
Edward C. Wente	1959
Georg von Békésy	1961
R. Bruce Lindsay	1963
Hallowell Davis	1965
Vern O. Knudsen	1967
Frederick V. Hunt	1969
Warren P. Mason	1971
Philip M. Morse	1973
Leo L. Beranek	1975
Raymond W. B. Stephens	1977
Richard H. Bolt	1979
Harry F. Olson	1981
Isadore Rudnick	1982
Martin Greenspan	1983
Robert T. Beyer	1984
Laurence Batchelder	1985
James L. Flanagan	1986
Cyril M. Harris	1987
Arthur H. Benade	1988
Richard K. Cook	1988
Lothar W. Cremer	1989
Eugen J. Skudrzyk	1990
Manfred R. Schroeder	1991
Ira J. Hirsh	1992
David T. Blackstock	1993
David M. Green	1994
Kenneth N. Stevens	1995
Ira Dyer	1996
K. Uno Ingard	1997
Floyd Dunn	1998
Henning E. von Gierke	1999
Murray Strasberg	2000
Herman Medwin	2001
Robert E. Apfel	2002
Tony F. W. Embleton	2002
Richard H. Lyon	2003
Chester M. McKinney	2004
Allan D. Pierce	2005
James E. West	2006
Katherine S. Harris	2007
Patricia K. Kuhl	2008
Thomas D. Rossing	2009
Jiri Tichy	2010
Eric E. Ungar	2011
William A. Kuperman	2012
Lawrence A. Crum	2013
Brian C. J. Moore	2014
Gerhard M. Sessler	2015
Whitlow W. L. Au	2016
William M. Hartmann	2017
William A. Yost	2018

CITATION FOR WILLIAM J. CAVANAUGH

". . . for practical applications to building design and education in architectural acoustics, and for service to the Society."

15 MAY 2019 • LOUISVILLE, KENTUCKY

Bill Cavanaugh has changed the world as we hear it. He has been at the forefront of a broad spectrum of research and consulting in architectural acoustics for over 60 years. His service to the Acoustical Society of America, to many related societies, and to the science and art of acoustics in general has been deep and pervasive. Although always eager to redirect credit to his colleagues, most of us know at least some of the roles he played that have helped to define the entire field of building acoustics.

William J. Cavanaugh and the Acoustical Society of America were both born in 1929–Bill in Boston. During World War II he attended The English High School, the first public high school in America, a prep school for students who had dreams of attending nearby Massachusetts Institute of Technology (MIT). Although Bill was too young to serve, during WWII, he told his father that he wanted to join the Army Corps of Engineers instead of entering the MIT freshman class in 1946. His father gave him that life changing advice only a father can give. "Bill why don't you just try it out for a year. If you don't like it, you can join the service." Bill entered MIT as a civil engineering major that summer where he met Walter Hill, an architecture student who asked Bill to help with a project which would be displayed in the Rotundra. This peaked his interest and Bill decided to switch his major to Architecture where none other than Robert Newman was one of his professors and mentors. Bill graduated in 1951 with a Bachelor in Architecture degree and was awarded the American Institute of Architects Student Medal. He received his commission as 2nd Lieutenant and was immediately ordered to active duty. where he served as a unit training and staff officer with the 6th Armored Division and U.S. Army Corps of Engineers, for the duration of the Korean Conflict. Shortly before the armistice was signed, Bill married Ginny Huff, who he met as a student, in 1953. He remained in the Army Reserves with the Army Corps of Engineers until 1982, retiring with the rank of full Colonel. He was awarded by the Army's Meritorious Service Medal in acknowledgement of his 31 ½ years of continuous service.

After his service, Bill joined the trailblazing acoustical consulting firm of Bolt Beranek and Newman (BBN) in February 1954, from which most acoustical consulting firms can trace a lineage, and where he was guided and deeply influenced by its famous partners, Dick Bolt, Leo Beranek and Robert Newman. Bob Newman, a remarkable and animated teacher, became Bill's immediate mentor, and encouraged Bill's own teaching, always emphasizing expressive language that would be persuasive to architects and engineers, and always with an irrepressible enthusiasm. Alongside his consulting practice, Bill soon began teaching acoustics classes at the MIT School of Architecture, the Boston Architectural College, the Rhode Island School of Design, Harvard School of Public Health and Rodger Williams College. Bill's inspirational teaching led many students to become future clients, and some even became acousticians. Bill once had Tony Hoover sit in on a few classes he taught at the Boston Architectural Centre and, to inspire his younger colleague said "OK, Tony you take over the class now."

Bill has been deeply involved in nearly all aspects of architectural acoustics and noise control consulting. He has consulted on thousands of projects of all building types, requiring skillful interaction with architects, engineers, and clients, as well as the public. He has a uniquely friendly, thorough, educational, and effective consulting style. Insights and research based on his projects have been the source of most of his contributions to the acoustical community at large. There are five areas of Bill's contributions to acoustics and its practical application that are worth special consideration; masking, outdoor venue sound propagation, cinema sound quality, the practice of architectural acoustics consulting, and teaching and mentoring.

Bill was the lead author on the seminal paper ''Speech Privacy in Buildings," J. Acoust. Soc. Am. 34, 475-492 (1962), that had the daring and the scientific evidence to suggest that adding appropriate background sound could improve acoustical privacy. More research and papers followed, leading to the entire industry of masking systems, and making open plan offices viable by increasing the potential for speech privacy, enhanced concentration, and decreased interruption. Speech privacy remains a high priority, especially, with new federal mandates for privacy in health-care facilities.

Bill's interest in the difficult problem of neighbors' complaints of sound from outdoor amphitheaters led to and has continued to influence the entire field of concert sound monitoring systems, associated methods for improved community relations, and development of acoustical criteria for outdoor concert venues. He has earned a reputation as a skilled negotiator, and sometimes peacemaker in difficult disputes.

Bill's work on cinemas during the 1980's and 1990's with so much activity in improving movie sonic quality led to extensive investigation and testing of movie theaters across the country, resulting in criteria that are now the foundation for quantifying appropriate sound isolation, HVAC noise, and finish treatments. This work had major influences on THX systems and certification of cinemas, which in turn have greatly affected the exploding home entertainment industry.

Bill's outreach to other acousticians and his optimistic vision of the future of acoustical consulting were instrumental in the formation, growth, and vigor of the National Council of Acoustical Consultants (NCAC) and the Institute of Noise Control Engineering (INCE). The benefits to consultants, architectural acousticians, and acoustics practice in general are ubiquitous, dramatic, and enduring. Bill served as President of the National Council of Acoustical Consultants (NCAC 1977-79), and was awarded their C. Paul Boner Medal for Distinguished Contributions to the Acoustical Consulting Profession in 1983, and the inaugural Laymon N. Miller Award for Excellence in Acoustical Consulting in 2015, awarded jointly by NCAC and INCE. He also served as President of the Institute of Noise Control Engineering (1993).

Bill's teaching and mentoring are of special interest. Bill is uniquely generous with his time and support to so many on various aspects of our own careers, research, and teaching. After 17 years with BBN, Bill established, Cavanaugh Tocci Associates, with ASA Fellow, Gregory Tocci., and is regularly represented at ASA, NCAC, and INCE meetings, and boasts three Past Presidents of the NCAC. He was a leader in establishing the Robert Bradford Newman Student Award Fund, a subcommittee of the ASA Technical Committee on Architectural Acoustics, which sponsors Newman Student Medal Awards for excellence in the study of architectural acoustics, as well as the Theodore J. Schultz Grants for excellence in teaching. Acoustics books and teaching aids have benefited from Bill's influence as well, including his essential role in originating the various ASA books based on poster sessions, including "Halls for Music Performance: Two Decades of Experience, 1962-1982," through "Halls for Music Performance: Another Two Decades of Experience, 1982-2002". Bill is the principal co-author of "Architectural Acoustics: Principles and Practice", published by Wiley (1st Ed. 1999, 2nd Ed. 2010). Bill's reputation has generated many invitations to write chapters, forewords, and articles on architectural acoustics and noise control, as well as deep participation in many Standards Working Groups.

Bill's service to the Society includes as three consecutive years as Member of the ASA Executive Council, member and Chair of the TCAA, member of the Technical Committee on Noise, and chair or member of numerous administrative committees including Ethics and Grievances, Regional Chapters, Public Relations, Rules and Governance, and more. In fact, Bill was the first Chair of the College of Fellows and of Archives and History. Bill served as Chair of the Wallace Clement Sabine Centennial Symposium in June 1994 and was responsible for the Historical Display at the Society's 50th Anniversary in 1979. He served on the 75th Anniversary Committee and was co-editor with Henry Bass of the ASA 75th Anniversary commemorative book. His work with local chapters and other organizations is outstanding. And, he was awarded the ASA Distinguished Service Citation in 1994, and the ASA Wallace Clement Sabine Award in December 2006.

For many years Bill has worked diligently, and often behind the scenes, for the benefit and recognition of others. He firmly believes, and his actions demonstrate, that much can be accomplished if no one cares who gets the credit. His wise and generous council has provided a bedrock foundation for the high standards for technical, personal, and professional conduct expected of practicing architectural acousticians. Bill's selfless outreach and mentoring to other consultants and acousticians has influenced many outstanding careers in acoustics.

Ginny, his wonderful, loving wife of 57 years, who passed peacefully in 2010, his five children (including Bill, Jr. who died in 1982 at age 24 while on active duty in the US Air Force), his grandchildren, and his great grandchildren are the loves of his life. He is also in love with acoustical consulting, and it shows.

His countless friends and admirers are proud to congratulate Bill on this richly-deserved ASA Gold Medal. Bill Cavanaugh, CONGRATULATIONS and THANK YOU!

K. ANTHONY HOOVER
LEO L. BERANEK
ALLAN D. PIERCE

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