Session 3aAA

Architectural Acoustics, Psychological and Physiological Acoustics, and Speech Communication: Auditory Perception in Virtual, Mixed, and Augmented Environments

Philip W. Robinson, Cochair
Media Technology, Aalto University, PL 15500, Aalto 00076, Finland

G. Christopher Stecker, Cochair
Hearing and Speech Sciences, Vanderbilt University, 1215 21st Ave. South, Room 8310, Nashville, TN 37232

Chair's Introduction—8:50

Invited Papers

8:55

3aAA1. Validating auditory spatial awareness with virtual reality and vice-versa. G. Christopher Stecker, Steven Carter, Travis M. Moore, and Monica L. Folkerts (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu)

"Immersive" technologies such as virtual (VR) and augmented reality (AR) stand to transform sensory and perceptual science, clinical assessments, and habilitation of spatial awareness. This talk explores some of the general challenges and opportunities for VR- and AR-enabled research, illustrated by specific studies in the area of spatial hearing. In one study, freefield localization and discrimination measures were compared across conditions which used VR to show, hide, or alter the visible locations of loudspeakers from trial to trial. The approach is well suited to understanding potential biases and cuing effects in real-world settings. A second study used headphone presentation to understand contextual effects on the relative weighting of binaural timing and intensity cues. Previous studies have used adjustment and lateralization to "trade" time and level cues in a sound-booth setting, but none have attempted to measure how listeners weight cues in realistic multisensory scenes or in realistic temporal contexts. Finally, a third study used simple VR "games" to measure implicit awareness of reverberation in multi-target scenes. Listeners' tolerance of reverberant mismatch can be used to assess and habilitate auditory spatial awareness and provides a testbed for future applications of auditory augmented reality.

9:15

3aAA2. Designing rehabilitative experiences for virtual, mixed, and augmented reality environments. Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Aaron Seitz (Psych., Univ. of California, Riverside, Riverside, CA), Timothy J. Vallier, and Dawna Lewis (Boys Town National Res. Hospital, Omaha, NE)

It is rapidly becoming possible to conduct nearly all of the diagnostic testing necessary for understanding suprathreshold auditory and visual processing abilities using consumer-grade electronics. It is also the case that all of the psychometric methods used are consistent with a game-based structure, which will encourage engagement and increase reliability of the testing. The obvious next step is to integrate diagnosis and rehabilitation into a single game-based framework, for which virtual, augmented, and mixed reality are an ideal platform. The possibilities include auditory, visual, and memory training and diagnostic games, as well as guided interactions to improve skills in real-life scenarios. Essential to effective design, however, is optimizing interfaces to facilitate universal access. Universal design is essential for rehabilitation because it is necessarily the case that those in need of services cannot be assumed to have optimal auditory, visual, tactile, and balance capabilities. The most exciting challenge for the future will involve designing environments that scale appropriately to engage, challenge, and improve all of the senses and cognitive systems of the participant while being flexible enough to provide engaging experiences for those who can most benefit from the experience.

9:35

3aAA3. Effects of non-individual head-related transfer functions and visual mismatch on speech recognition in virtual acoustic environments. Kristi M. Ward (Northwestern Univ., 2240 N. Campus Dr., Rm. 2-381, Evanston, IL 60208, kmward@u.northwestern.edu), Z. Ellen Peng (Univ. of Wisconsin–Madison, Madison, WI), and Tina M. Grieco-Calub (Northwestern Univ., Evanston, IL)

There is widespread research and clinical interest in quantifying how the acoustics of real-world environments, such as background noise and reverberation, impede a listener's ability to recognize speech. Conventional methods used to quantify these effects include dichotic listening via headphones in sound-attenuated booths or loudspeakers in anechoic or low-reverberant environments, which lack the capability of manipulating room acoustics. Using a state-of-the-art Variable Room Acoustics System housed in a virtual sound room (ViSoR), this study aims to systematically assess the effects of non-individual head-related transfer functions (HRTFs) and mismatched visual perception on speech recognition in virtual acoustic environments. Young adults listened to and repeated sentences presented amidst a co-located two-talker speech competitor with reverberation times ranging from 0.4 to 1.25 s. Sentences were presented in three listening conditions: through a loudspeaker array in ViSoR with the participants' own HRTFs (*Condition 1*); via headphones in a sound-attenuated booth with non-individual HRTFs(*Condition 2*); and using the same binaural reproduction as *Condition 2* in ViSoR (*Condition 3*). *Condition 3* serves as a control condition, allowing us to quantify the separate effects of non-individual HRTFs and visual mismatch on speech recognition. Discussion will address the validity and use of virtual acoustics in research and clinical settings.

9:50

3aAA4. Use of non-individualized head-related transfer functions to measure spatial release from masking in children with normal hearing. Z. Ellen Peng, Ruth White, Sara Misurelli, Keng Moua, Alan Kan, and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53711, zpeng49@wisc.edu)

Spatial hearing studies with children have typically been conducted using loudspeakers in laboratories. However, loudspeaker arrays are rare in clinics due to high cost and technical set-up requirements. The use of virtual auditory space (VAS) with non-individualized head-related transfer functions (HRTFs) can increase the feasibility of assessing spatial hearing abilities in clinical settings. A novel paradigm for measuring spatial release from masking (SRM) was developed using non-individualized HRTFs. This paradigm measures the minimum angular separation needed between target and masker to achieve a 20% increase in target speech intelligibility. First, the 50% speech reception threshold (SRT) was measured with target and masker co-located to one side. Then, the masker position was adaptively changed to achieve 70.7% intelligibility while maintaining the signal-to-noise ratio at the level of the co-located SRT. To verify the use of non-individualized HRTFs, normal-hearing children were tested (1) using a loudspeaker array and (2) in headphone-based VAS created using KEMAR HRTFs measured in the same setup as (1). Preliminary results showed that co-located SRTs and target-masker angle separation to achieve a 20% SRM were similar in loudspeaker array and in headphone-based VAS. This suggests that non-individualized HRTFs might be used in an SRM task for clinical testing.

10:05

3aAA5. How physical versus panned sources in dry or reverberant conditions affect accuracy of localization in sound field synthesis systems. Anna C. Catton, Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, anna.catton@huskers.unl.edu), Adam K. Bosen, Timothy J. Vallier, and Douglas H. Keefe (Boys Town National Res. Hospital, Omaha, NE)

Sound field synthesis systems vary in number and arrangement of loudspeakers and methods used to generate virtual sound environments to study human hearing perception. While previous work has evaluated the accuracy with which these systems physically reproduce room acoustic conditions, less is known on assessing subjective perception of those conditions, such as how well such systems preserve source localization. This work quantifies the accuracy and precision of perceived localization from a multi-channel sound field synthesis system at Boys Town National Research Hospital, which used 24 physical loudspeakers and vector-based amplitude panning to generate sound fields. Short bursts of broadband speech-shaped noise were presented from source locations (either coinciding with a physical loudspeaker location, or panned between loudspeakers) under free-field and modeled reverberant-room conditions. Listeners used a HTC Vive remote laser tracking system to point to the perceived source location. Results on how physical versus panned sources in dry or reverberant conditions impact accuracy and precision of localization are presented. Similar validation tests are recommended for sound field synthesis systems at other labs that are being used to create virtual sound environments for subjective testing. [Work supported by NIH GM109023.]

10:20–10:35 Break

10:35

3aAA6. Evaluating immersion of spatial audio systems for virtual reality. Lukas Aspöck, Michael Kohnen, and Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de)

Multiple aspects influence the quality of experience of different VR presentations. One popular aspect which is usually considered for rating the presentation is the concept of immersion. Its complex nature as well as indistinct definitions make it challenging to use it as an objective measure in scientific experiments. Additionally previous studies revealed contradictory definitions of immersion and its separation from the concept of presence. To investigate immersion, a nomological net was developed which connects various items contributing to immersion. These items were assigned to subcategories and should ideally be well defined and measurable. For each item, multiple questions were formulated. Pre-testing on their linguistic quality and unambiguity was conducted to identify suitable questions for each item. These questions were applied in two listening experiments: A between-group design for the evaluation of the chosen questions and a within-subject design for the evaluation of differences between Higher-Order Ambisonics, VBAP, and binaural loudspeaker reproduction.

10:50

3aAA7. Perceptually plausible room acoustics simulation including diffuse reflections. Oliver Buttler (Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Oldenburg, Germany), Torben Wendt, Steven van de Par (Acoust. Group and Medizinische Physik and Cluster of Excellence Hearing4all, Universität Oldenburg, Oldenburg, Germany), and Stephan D. Ewert (Medical Phys. and Cluster of Excellence Hearing4all, Universität Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, Stephan.ewert@uni-oldenburg.de)

Immersive and convincing acoustics in virtual reality applications require computationally efficient methods. The fast and perceptually plausible room acoustics simulator [RAZR, see Wendt *et al.*, JAES, 62, 11 (2014)] approaches this demand by drastic simplifications with respect to physical accuracy while still accomplishing perceptual plausibility. RAZR is based on a hybrid approach where early reflections are calculated as image sources for a shoebox-room approximation up to a low order, and the later reverberation is generated by a binaurally extended feedback-delay-network (FDN). Although it was demonstrated that a good perceptual agreement with real non-shoebox rooms is achieved, the shoebox-room simplification might cause limitations for rooms which strongly diverge from this assumption. Here the perception of temporal smearing of early diffuse reflections, effectively simulating effects of scattering and multiple reflections caused by geometric disturbances at walls and by objects in the room, was systematically assessed. A single parameter was introduced to quantify

deviations from an empty shoebox room. It was demonstrated that perceptually plausible results can be obtained for auralized natural stimuli and for the binaural impulse responses themselves. It is shown how parameters in RAZR are derived from room geometry and surface materials or from measured BRIRs or frequency-dependent reverberation times.

11:05

3aAA8. Evaluation of near field distance perception in virtual environments. Philip W. Robinson (Oculus Res., PL 15500, Aalto 00076, Finland, philrob22@gmail.com)

Binaural synthesis of spatial audio for virtual and augmented environments is typically performed by convolving an anechoic signal with impulse responses measured at the listener's ears using a source at a fixed distance. This method is accurate for sounds presented at the measured impulse response's distance, and for more distant sources, decreasing intensity provides a reasonable approximation. However, sources nearer than 1 m provide additional distance cues. The head-shadow effect becomes markedly stronger and exponential sound intensity falloff comes into play, exaggerating inter-aural level differences. Also, the angle to each ear diverges, changing the directionally dependent spectral filtering of the pinnae. Changes in inter-aural level differences can be approximated across listeners, but changes in pinna cues with changes in distance are highly individual, and thus less easily approximated. Reproducing some or all of these cues for each individual may be necessary to create a convincing percept of very near objects in virtual and augmented reality environments. The present work aims to determine the importance of each cue for static and dynamic sources. Listening tests have been conducted in a virtual environment using generic and individual head related impulse responses, with and without near-field compensation.

Invited Papers

11:20

3aAA9. Improving the perception of a sound source's polar angle in mixed reality. Hannes Gamper and Ivan J. Tashev (Res., Microsoft, One Microsoft Way, Redmond, WA 98052, hannes.gamper@microsoft.com)

Mixed reality applications blend real and virtual scenes. To render virtual objects in a scene, the rendering system needs to accurately control their perceived location. In the acoustic domain, the location of a sound source is often given in interaural coordinates, i.e., as the lateral and polar angle and distance relative to the midpoint of the listener's interaural axis. This description is useful as it allows separating the effect of various perceptual cues on each interaural spatial dimension. Prior research has shown that the human perception of a sound source's polar angle, i.e., the angle of rotation about the interaural axis, tends to be less accurate than the perception of its lateral angle, i.e., the angle off the median plane. When rendering virtual sound sources, listeners often confuse locations above and below the horizontal plane or in the front and in the back. Here, we review cues that affect the perception of polar angle as well as approaches to improve the accuracy of rendering the polar angle of a virtual sound source in mixed reality applications.

11:40

3aAA10. Localization and externalization in binaural reproduction with sparse HRTF measurement grids. Zamir Ben-Hur (Oculus & Facebook and Ben-Gurion Univ. of the Negev, Be'er Sheva, Israel), David L. Alon (Oculus & Facebook, Menlo Park, CA), Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva, Israel), and Ravish Mehra (Oculus & Facebook, 8747 148th Ave. NE, Redmond, WA 98052, ravish.mehra@oculus.com)

High-quality spatial sound reproduction is important for many applications of virtual and augmented reality. A key component in spatial audio reproduction is the head-related transfer function (HRTF). To achieve a realistic spatial audio experience, in terms of sound localization and externalization, high resolution individualized HRTFs are necessary. However, these are typically unavailable as they require specialized equipment and a large number of measurements, which motivates the use of sparsely measured HRTFs. Reducing the number of measurement points requires spatial interpolation, and may lead to spatial aliasing error. Previous studies suggested the use of spherical-harmonics (SH) decomposition for the interpolation. With a sparse grid, the SH representation is order-limited, leading to a constrained spatial resolution due to the truncation error. In this study, the effect of sparse measurement grids on the reproduced binaural signal is perceptually evaluated, through localization and externalization tests under varying conditions of aliasing and truncation. Preliminary results indicate relatively large effect of SH order on these attributes, while smaller effect is observed due to aliasing error.

Session 3aAO

Acoustical Oceanography, Animal Bioacoustics, and Underwater Acoustics: Ambient Noise Oceanography in Polar Regions: Noise Properties and Parameter Estimation

Oskar Glowacki, Cochair
Institute of Geophysics Polish Academy of Sciences, Ksiecia Janusza 64/413, Warsaw 01-452, Poland

Grant B. Deane, Cochair
Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St, La Jolla, CA 92093-0238

Chair's Introduction—8:15

Contributed Paper

8:20

3aAO1. Characteristics of the under-ice soundscape in the southern Beaufort Sea during Ice Exercise 2016. John E. Joseph, D. Benjamin Reeder, and Tetyana Margolina (Oceanogr., Naval Postgrad. School, 833 Dyer Rd, Monterey, CA 93943, jejoseph@nps.edu)

Ice Camp SARGO was the remote hub of operations for the multi-national naval operation Ice Exercise 2016 (ICEX-16), held in March of that year. Over a three-day period in early March, continuous recordings of the under-ice soundscape were collected with receivers deployed at various depths through first-year ice in the vicinity of the ice camp as it drifted westward across the Beaufort Sea approximately 175 nm north of Prudhoe Bay, Alaska. A significant reduction in the strength of easterly winds resulted in deceleration of the ice sheet during the period of observation, inducing notable ice cracking and ridging events near the camp. Ice sheet movement slowed from about 0.5 knots early in the test to virtually coming to a halt near the end of the recording period. Sounds from naturally occurring and anthropogenic sources in the 10-Hz to 10-kHz band detected in the recordings were analyzed in connection to the origins of the sound and correlated to varying environment conditions including wind speed and ice motion. Results show a wide variety of persistent and transient sound sources contribute to the total soundscape at this remote location. Comparisons to other soundscape observations in the region are discussed.

Invited Paper

8:35

3aAO2. Underwater noise measurements during a year long Shallow Water Canada Basin Acoustic Propagation experiment from 2016 to 2017. Mohsen Badiey (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu), Ying-Tsong Lin (Woods Hole Oceanographic Inst., Woods Hole, MA), Sean Pecknold (DRDC Atlantic Res. Centere, Dartmouth, NS, Canada), Altan Turgut (Naval Res. Lab., Washington, DC), Megan S. Ballard, Jason D. Sagers (Appl. Res. Labs., Austin, TX), and Christopher Whitt (JASCO Appl. Sci., Dartmouth, NS, Canada)

A year-long, multi-institution, acoustical oceanographic measurement on the Chukchi Continental Shelf region of the Canada Basin started on October 2016 is reported. Ten vertical receiver line arrays and a horizontal receiver line array were deployed together with oceanographic sensors to measure the sound speed profiles, ice formation, and currents over an area of approximately 30 km by 50 km. Various aspects of the noise including spectral fluctuations, directionality, seasonal dependence, and intensity fluctuations are studied over time. In this paper, we present preliminary measurement results depicting spatial and temporal distribution of underwater background noise at the experiment site. [Work supported by ONR321 OA.]

Contributed Papers

8:55

3aAO3. Shallow water propagation of mid-frequency cross-shelf acoustics on the Chukchi Shelf. Justin Eickmeier, Mohsen Badiey (Univ. of Delaware, 17 McCord Dr., Newark, DE 19713, jeickmei@udel.edu), and Altan Turgut (Acoust. Div., Naval Res. Lab, Washington, DC)

The shallow water Canada Basin Acoustic Propagation Experiment (SW CANAPE) was conducted to study the effects of oceanographic variability on broadband acoustic fields in the Arctic. The physics of the acoustic waveguide on the northeastern edge of the Chukchi Shelf are influenced by dynamic boundary conditions and spatio-temporal fluctuations in the water column temperature and salinity profiles. Several oceanographic and acoustic receiving arrays were deployed across the Chukchi Shelf out to the shelf break region. Linear frequency modulated (LFM) signals were transmitted by two sources on the shelf for a long period of time. The influence of small scale, short-term water column variability, and dynamic upper boundary conditions including open water, marginal, and solid ice zones on shallow water propagation is shown for a 10 km source-receiver separation with well-defined water column properties measured at the source, receiver, and a mid-point along the cross-shelf acoustic path. [Work supported by ONR 321OA.]

9:10

3aAO4. Properties of the ambient noise field recorded at the 150 m isobath during the 2016–2017 Canadian Basin Acoustic Propagation Experiment. Jason D. Sagers and 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)

The Applied Research Laboratories at the University of Texas at Austin (ARL:UT) deployed two passive acoustic recording systems along the 150 m isobath of the Chukchi Shelf during the 2016–2017 Canadian Basin Acoustic Propagation Experiment (CANAPE). The first system was a single-hydrophone recorder located on the seafloor, while the Persistent Acoustic Observation System (PECOS) contained a horizontal line array of hydrophones along the seabed and a vertical line array spanning a portion of the water column. The systems were deployed and recovered during open-water conditions, but remained in place during the ice-formation, ice-covered, and ice-melt time periods. This talk presents initial findings of the statistical ambient noise levels during the year-long experiment, presents beam-noise levels recorded by PECOS, and qualitatively discusses the natural, biologic, and anthropogenic sounds present in the acoustic recordings. [Work sponsored by ONR.]

9:25–9:40 Break

Invited Paper

9:40

3aAO5. Acoustical characteristics and contributions of bubbles released from melting glacier ice. Matthew C. Zeh, Preston S. Wilson, Kevin M. Lee (Dept. Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 3374 Lake Austin Blvd., Apt. B, Austin, TX 78703, mzeh@utexas.edu), and Erin C. Pettit (Dept. of GeoSci., Univ. of Alaska Fairbanks, Fairbanks, AK)

Glacierized fjords present a unique acoustic environment that is significantly louder than other ice-covered environments, with average sound pressure levels of 120 dB re 1 μPa with a broad peak between 1 and 3 kHz [Pettit *et al*., Geophys. Res. Lett. **42**, 2309–2316 (2015)]. The energy within this peak is due to the release of bubbles escaping from pressurized air-filled pores within melting glacier ice. These bubbles form during glacier formation via the compression of successive seasonal snow layers. During ice melt, the pressurized air cavities are released, jetting and squirting from the ice and into the surrounding ocean environment [Lee *et al*., Proc. Meetings Acoust. **20**, 070004 (2014)]. In order to more sufficiently characterize this dynamic physical process, acoustic measurements and high-speed video recordings were made using melting glacier ice samples taken from LeConte Glacier in southeastern Alaska. Samples were melted and recorded in a controlled laboratory apparatus as well as at a range of depths at Lake Travis northwest of Austin, TX. Insights gained from these tests are then compared to field recordings taken in LeConte Bay from October 2016 to September 2017. [Work supported by NDSEG Fellowship and ONR.]

Contributed Papers

10:00

3aAO6. Trends in wind, rain, and ships measured from ambient sound in the Bering Sea. Jennifer L. Miksis-Olds (School of Marine Sci. & Ocean Eng., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, j.miksisolds@unh.edu) and Jeffrey Nystuen (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Advances in low-cost passive acoustic technology are providing long term, ambient sound time series from which to better understand the complex interactions between marine life, the environment, and mankind in remote, seasonally ice covered areas such as the Bering Sea. Remotely deployed Passive Acoustic Listeners (PALs) operate according to an adaptive sampling protocol as opposed to recording continuously, thus enabling data collection for a full year at a sampling rate of 100 kHz. Adaptive subsampling overcomes hardware and power limitations by reducing the amount of data acquired, while maintaining a high probability of detection and classification of targeted signals. Source contributions to the overall soundscape vary in space and time, and are highly dependent on temporal, geographic, and oceanographic factors. The differences observed in the acoustic record over time and between locations reflect changes in both physical and biological dynamics. This study focused only on the changing patterns of seasonal sea ice dynamics, storm frequency and intensity, and human activity in the acoustic record. As polar conditions continue to change due to reductions in sea ice, warming ocean temperatures, and increases in human use, passive acoustic monitoring will be a valuable tool to identify and track weather dynamics.

10:15

3aAO7. Studying melting icebergs with ambient noise oceanography. Oskar Glowacki (Inst. of Geophys. Polish Acad. of Sci., Ksiecia Janusza 64/413, Warsaw 01-452, Poland, oglowacki@igf.edu.pl), Grant B. Deane (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA), and Mateusz Moskalik (Inst. of Geophys. Polish Acad. of Sci., Warsaw, Poland)

Marine-terminating glaciers are retreating at an unprecedented pace, largely as a result of enhanced submarine melting. However, studying ice-ocean interactions is complicated due to both harsh conditions prevailing in glacial bays and lack of scientific methods enabling continuous measurements. Recent studies have shown that high underwater noise levels measured in the Arctic are related to glacier melt, but quantitative research requires proper separation of individual noise sources, including icebergs and glacier fronts. Therefore, we show results of field experiments carried out in 2013, 2015, and 2016 in Hornsund fjord, Svalbard, to present directionality and statistics of the noise produced by melting icebergs. Measurements of noise directionality were conducted with 3-hydrophone acoustic array. Calculated angles of arrivals for the noise at the frequency range of 1–10 kHz correspond well to locations of individual, grounded icebergs. The amplitude of sound emitted by these sources has a symmetric α-stable distribution, with parameters depending on separation between iceberg and receiver. These findings demonstrate that ambient noise oceanography is an efficient tool to detect and track icebergs using natural noise they produce during melting. [Work funded by the Polish National Science Centre, Grant No. 2013/11/N/ST10/01729 and by ONR, Grant No. N00014-17-1-2633.]

10:30

3aAO8. The vertical directionality of melt noise from a glacier terminus. Grant B. Deane (Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St, La Jolla, CA 92093-0238, gdeane@ucsd.edu) and Oskar Glowacki (Polar Sci., Inst. of Geophys., Warsaw, Poland)

Measurements of the vertical directionality of the sound of bubbles released explosively from a melting glacier terminus are presented. These data are motivated by a desire to infer ice melt rates using the sound generated by bubbles, trapped in the glacier ice at the base of the fern layer and pressurized over time, as they are released by the melting terminus. The free energy available to generate noise is a function of the difference between the bubble internal gas pressure and hydrostatic pressure, both of which can vary with depth beneath the sea surface. Previous studies of bubble gas pressure in glacier ice suggest that the noise generated by bubble release should decrease with increasing depth below the surface. Measurements of noise directionality made with a compact, 4-element hydrophone array approximately 300 m in front of Hansbreen Glacier in Hornsund Fjord, Southwestern Svalbard in the summer of 2017 will be presented and discussed. This study suggests that only the top few 10's of meters of ice cliff generate the majority of melt noise. This result has important implications for the interpretation of bubble noise in terms of ice melt rate. [Work funded by ONR, Grant No. N00014-17-1-2633 and by the Polish National Science Centre, Grant No. 2013/11/N/ST10/01729.]

WEDNESDAY MORNING, 9 MAY 2018 GREENWAY F/G, 9:00 A.M. TO 12:00 NOON

Session 3aBA

Biomedical Acoustics and Physical Acoustics: Induction Mechanisms for Bubble Nucleation I

Jeffrey B. Fowlkes, Cochair
Radiology, Univ. of Michigan, 3226C Medical Sciences Building I, 1301 Catherine Street, Ann Arbor, MI 48109-5667

Ronald Roy, Cochair
Engineering Science, University of Oxford, Parks Road, Oxford OX1 3PJ, United Kingdom

Chair's Introduction—9:00

Invited Papers

9:05

3aBA1. Homogeneous vs heterogeneous nucleation, a comparison. Charles C. Church (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, cchurch@olemiss.edu)

Vapor bubbles of sufficient size to serve as effective acoustic scatterers or cavitation nuclei may form in materials having regions of very low interfacial tension when exposed to an acoustic wave. These regions may be simple "hot spots," *i.e.*, collections of molecules having higher-than-average kinetic energies which therefore have lower surface tension than is found a room temperature, or the interfaces between dissimilar materials, *e.g.*, the lipid coating on a microdroplet of liquid perfluourocarbon used as an ultrasound contrast agent. In addition, the curvature of such interfaces can act to focus an impinging acoustic beam in such a way as to increase its rarefactional pressure above that of the incoming wave. Another factor not often considered is that any foreign material, whether dissolved liquid or minute solid particulate, can act to disrupt the normal molecular structure of the liquid, thereby providing local regions of reduced interfacial tension. The energetics of the nucleation process shows that the rate of nucleation events is highly sensitive to the instantaneous temperature and moderately sensitive to both the magnitude and the duration of the rarefactional pressure of the acoustic wave. These and additional factors influencing nucleation processes will be discussed.

9:25

3aBA2. Histotripsy—From nucleation of bubbles to tissue disruption. Jeffrey B. Fowlkes (Radiology, Univ. of Michigan, 3226C Medical Sci. Bldg. I, 1301 Catherine St., Ann Arbor, MI 48109-5667, fowlkes@umich.edu)

Histotripsy utilizes focused ultrasound to nucleate gas bubbles deep within tissue and the associated bubble activity to induce biological effects at the level of tissue disruption/emulsification. Ultrasonic fields for inducing such effects include both pulses with as little as a single predominant negative half cycle to pulses with thousands of cycles and more. Different ensembles of bubbles leads to differing proposed mechanisms for tissue damage. Control of the damaged area is due to the threshold phenomena associated with the nucleation and the large focal gain in the main lobe, where the tissue disruption can be confined to fractions of the lateral beam width with little collateral damage. In addition, manipulation of the residual nuclei from a given ultrasound pulse can control cavitation events in subsequent pulses to allow greater and more uniform effects. The degree of tissue damage is equally impressive where disruption can be down to the subcellular level. At the same time, ultrasound parameters can be adjusted to selectively spare larger vessels from damage or retain the extracellular matrix. This variety of histotripsy properties opens up numerous potential medical applications, which will be discussed along with the conditions needs to achieve controlled and effective tissue disruption.

9:45

3aBA3. Nucleation and stabilization of cavitation nuclei in liquids. Lawrence A. Crum (APL, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, lacuw@uw.edu) and Julianna C. Simon (Acoust., Penn State University, University Park, PA)

The presence of cavitation nuclei are normally required in order to induce cavitation in water-based fluids, as the homogeneous nucleation threshold is beyond the range of most acoustic-pressure generation systems (a notable exception is the "intrinsic threshold" achieved by some histotripsy devices). There are a number of potential models for these nuclei, but one that continues to receive favor is the "crevice model," in which a pocket of gas is contained in a crack or crevice in a contaminating particle and then stabilized against diffusion by the geometry of the crevice. This presentation will describe this model in some detail, present some (old) data that support the model, and some new data that imply that such activatable nuclei exist in mammalian tissue.

10:05

3aBA4. The induction of inertial cavitation from polymeric nanocups—From theory to observation. James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., 62 Nanyang Dr., Block N1.2, 01-06, Singapore 637459, Singapore, james-kwan@ntu.edu.sg), Guillaume Lajoinie (Phys. of Fluids, Univ. of Twente, Enschede, Netherlands), Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Michel Versluis (Phys. of Fluids, Univ. of Twente, Enschede, Netherlands), and Constantin Coussios (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

The inability for therapeutics to distribute throughout the entirety of the tumor is a major challenge in cancer therapy. Acoustic cavitation from microbubbles promotes drug distribution and improves therapeutic efficacy. Yet microbubbles are too large to navigate the microvasculature of the tumor, and are destroyed by the ultrasound wave. Thus, there is a need for submicron cavitation nucleation agents. Recently, we have developed a submicron polymeric nanocup capable of trapping a nanobubble within the surface crevice. Using a modified Rayleigh-Plesset model that accounts for the size and shape of the crevice on the surface of the nanocup, we predicted that the cavity trapped bubble expands and contracts yet remains in the cavity. With a sufficient peak negative pressure amplitude, the model indicated that the surface bubble extends beyond and detaches from the crevice before inertial collapse. To verify our predictions, we exposed nanocups to high intensity focused ultrasound at different driving frequencies and observed bubble nucleation from nanocups using the Brandaris ultra high-speed camera. Acoustic emissions were recorded using a passive cavitation detector. These direct observations of the induction of inertial cavitation from nanocups verified the predictions made by the modified Rayleigh-Plesset crevice model.

10:25–10:40 Break

10:40

3aBA5. The effect of hypobaric pressure on the kidney stone twinkling artifact. Julianna C. Simon (Graduate Program in Acoust., The Penn State Univ., Univ. Park, PA and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA and Dept. of Acoust., Moscow State Univ., Moscow, Russian Federation), Wayne Kreider, Michael Breshock (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), James C. Williams (Dept. of Anatomy and Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN), and Michael Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Recently, our group proposed the color Doppler ultrasound twinkling artifact originates from stable crevice bubbles on the kidney stone surface because overpressure suppressed twinkling on *ex vivo* stones (Lu *et al., Ultrasound Med. Biol.* 2013). However, the hypothesis is not fully accepted because the bubbles were not directly observed. Here, the submicron-sized bubbles predicted by the crevice-bubble hypothesis are enlarged in *ex vivo* kidney stones by exposure to a pre-focal, off-axis lithotripter pulse (p_+ = 1.5 MPa, p_-=2.5 MPa) or hypobaric static pressures (0.021 MPa, absolute) to simultaneously capture their appearance by high-speed photography and ultrasound imaging. In rough stones that twinkle, consecutive lithotripter pulses caused more than 50% of bubbles to grow reproducibly from specific locations on the stone surface, suggesting the bubbles were pre-existing. Conversely, on smooth stones that did not twinkle, repeated lithotripter pulses initiated bubbles from varying locations on the stone surface. Similarly, upon exposure to hypobaric static pressures, the simple expectation that twinkling would increase by enlarging bubbles largely held for rough-surfaced stones but was inadequate for smoother stones. These results suggest a correlation between kidney stone surface topography or stable surface crevice bubbles and twinkling. [Work supported by NSBRI through NASA NCC 9-58 and NIH DK043881.]

11:00

3aBA6. Laser nucleation of single bubbles and clouds in an acoustic resonator via pressure-dependent dielectric breakdown. R. G. Holt (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, rgholt@bu.edu), Jonathan R. Sukovich (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), Phillip Anderson (Listen, Inc., Boston, MA), Ashwinkumar Sampathkumar (Dayton Res. Ctr., Riverside Res., New York, New York), Todd W. Murray (Mech. Eng., Univ. of Colorado, Boulder, CO), and D. F. Gaitan (Sound Eng., Univ. of San Buenaventura, Medellin, Colombia)

Obtaining bubbles on demand at precise times and locations in a non-contact fashion can be useful in a variety of applications. Of special importance is the combination of laser nucleation with acoustics, so that bubbles are only just nucleated by the optics but grown to macroscopic size solely by the acoustics. We present theory and experiment for the non-thermal laser nucleation of bubbles in an acoustic field in the absence of significant absorbing/scattering particles. First we present theory and experiment for the threshold for dielectric breakdown in water, resolving the distinct minimum at 20 bar. Then, we present a method and results for nucleating single and multiple bubbles with temporal uncertainty of 5 ns, and spatial uncertainty of 1 mm. Results for bubble number and first cycle expansion are reported as functions of the timing of the nucleating laser pulse with respect to the acoustic field. [Work supported by Impulse Devices, Inc.]

11:20

3aBA7. Micro-cavitation on demand via the nanoparticle mediated interaction of light and sound. Ronald Roy (Dept. of Eng. Sci., Univ. of Oxford, Parks Rd., Oxford OX1 3PJ, United Kingdom, ronald.roy@hmc.ox.ac.uk), Caleb H. Farny (Mech. Eng., Boston Univ., Boston, MA), Tianming Wu (Radiation Oncology, Univ. of Chicago, Chicago, IL), R. G. Holt (Mech. Eng., Boston Univ., Boston, MA), and Todd W. Murray (Mech. Eng., Univ. of Colorado, Boulder, CO)

The safe utilization of controlled cavitation for HIFU therapy and ultrasound assisted drug delivery requires nucleation sites for bubble formation. We consider the potential for nucleating transient vapor cavities using laser-illuminated gold nanoparticles combined with high-intensity focused ultrasound. An transparent polyacrylamide gel phantom was seeded with 82-nm diameter gold particles and exposed to 20 ns pulses from a 532 nm Nd:Yag laser. Laser firing was synchronized with the arrival of a burst of 1.1 MHz focused ultrasound. Acoustic emissions from ensuing inertial cavitation were detected passively using a 15 MHz focused transducer. At a laser energy of 0.10 mJ/pulse, the resulting inertial cavitation nucleation threshold pressure (peak-negative focal pressure) was as low as 0.92 MPa. In comparison, a peak-negative focal pressure of 4.50 MPa was required to nucleate detectable cavitation without laser illumination (nano-particles were present in both cases). Experimental results agree well with a simple model for transient heating and cavity formation. Since the particles are durable, one can re-activate them as needed, essentially yielding cavitation nuclei "on demand." [Work supported by the Dept. of the Army (Award No. DAMD17-02-2-0014) and the Center for Subsurface Sensing and Imaging Systems (NSF ERC Award No. EEC-9986821).]

11:40

3aBA8. Nucleation pressure threshold in acoustic droplet vaporization. Christopher Miles, Charlie Doering, and Oliver D. Kripfgans (Dept. of Radiology, Univ. of Michigan, Ann Arbor, MI 48109-5667, oliver.kripfgans@umich.edu)

Acoustic droplet vaporization (ADV) has been studied by several laboratories around the world, including the parametric dependence of its nucleation. In this abstract, we will present our approach of combining classical homogeneous nucleation theory with superharmonic focusing to predict necessary pressures to induce nucleation in ADV. We will show that linear acoustics is a valid approximation to leading order when particle displacements in the sound field are small relative to the radius of the exposed droplet. This is done by perturbation analysis of an axisymmetric compressible inviscid flow about a micrometer-sized droplet with small surface perturbations relative to the mean droplet radius when subjected to an incoming ultrasonic wave. A calibrated single element spherical focus transducer (Olympus A321S 7.5 MHz) was positioned in a water tank above an inverted microscope (Leica DM IL) to emit single ten-cycle bursts. Single droplets (3–30 μm) were placed on ultrasound gel and exposed quasi free-field. Theoretical derivations and predictions will be compared to experimental findings. The ADV nucleation pressure threshold inside the droplet is calculated to be -9.33 ± 0.30 MPa for typical experimental parameters. As a result we are able to predict if a given incident pressure waveform will induce nucleation in an exposed perfluorocarbon droplet.

Session 3aEAa

Engineering Acoustics: General Studies on Structures

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

8:15

3aEAa1. Application of acoustic fabrics to improve the insertion loss of a partial enclosure. Weiyun Liu and David Herrin (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, weiyun.liu@uky.edu)

The sound absorption coefficient, transmission loss, and transfer impedance of different sound absorbing fabrics are measured. It is shown that the properties are similar to microperforated panels. Using microperforated panel equations, effective hole diameter and porosity are determined via a least squares curve fit. The effective parameters can then be used to predict the sound absorptive performance for different cavity depths. It is demonstrated that the fabrics have good sound absorptive performance over a wide range of frequencies. The fabric was then positioned in a partial enclosure. Single and double layers of the fabric were placed over an opening and the insertion loss as a result of fabric application was measured. The insertion loss is increased by approximately 5 dB if two layers of the fabric is used.

8:30

3aEAa2. Determination of the inverse, blocked, and pseudo forces for an air compressor bolted to a steel plate. Keyu Chen and David Herrin (Dept. of Mech. Eng., Univ. of Kentucky, 151 Ralph G. Anderson Bldg., Lexington, KY 40506-0503, keyu.chen@uky.edu)

For many machines, input forces cannot be directly measured, but representative forces may be determined using inverse approaches. Three approaches have been suggested in the literature for the determination of representative forces. The different approaches are described and then applied to an air compressor bolted to a steel plate. The three approaches are utilized to determine the inverse, blocked, and pseudo forces, respectively. The calculated indirect forces are then used to determine the response on the steel plate, and there is a good agreement between all three methods. The calculated forces are then used to predict the effect of a boundary condition modification to the steel plate.

8:45

3aEAa3. Analytical and experimental methods for the non-destructive acoustic testing of fluid-filled critical infrastructure from the 19th century to the present. Harrison Richarz (ICONAC / VESI Boesman, 76 Roxborough Ln., Thornhill, ON L4J 4T4, Canada, harrison@iconac.co)

From boreholes and chimneys, to pipelines, boilers, storage tanks, reactors, and water mains, non-destructive acoustic tests of fluid-filled conduits and shells help to monitor vital infrastructure across the globe. The theoretical treatment of wave propagation in such structures has its origin in the 19th century. In the intervening years, advances in sensors, data logging, and signal processing led to practical applications, first in oil exploration and later in other fields such as medicine and water distribution. This commercial activity accelerated the understanding of wave propagation in large scale multi-layered and multimaterial structures to create a host of new commercial applications. The paper explores the evolution and current state of these technologies and the acoustics behind them through an historical study of the progression of conduit wall impedance modeling and the problems of reflection and dispersion, as encountered in borehole and pipe inspection applications.

9:00

3aEAa4. Spatiotemporal modulation for mid-air haptic feedback from an ultrasonic phased array. Brian Kappus (Ultrahaptics, 2479 East Bayshore Rd., Palo Alto, CA 94303, brian.kappus@ultrahaptics.com) and Ben Long (Ultrahaptics, Bristol, United Kingdom)

<!–StartFragment–>A tactile sensation can be experienced by focusing airborne ultrasound using a phased array. Nonlinear acoustic pressure alone is difficult to perceive so traditional methods use amplitude modulation in the range of 10–300 Hz to stimulate nerves on the hand most sensitive to those frequencies. We demonstrate that through rapid translation of focus points similar results can be obtained using spatiotemporal modulation. This allows for volumetric sensations to be created using maximum power possible from the array. This is made possible through a solving approach using a pseudo-inverse of the activation matrix followed by a power iteration eigenvalue solution. Advantages versus amplitude modulation and consequences for parametric audio will be discussed.<!–EndFragment–>

9:15

3aEAa5. Boundary condition effects of porous material in absorption measurements: Comparison of two impedance tubes. Bárbara Fengler, William D. Fonseca, Paulo Mareze, Eric Brandao (Acoust., UFSM, Santa Maria, RS, Brazil), and ARTUR Zorzo (Acoust., UFSM, SQN 315 BLG Apto 205, Asa Norte, Brasilia 70774070, Brazil, arturzorzo@hotmail.com)

Porous materials are widely used for acoustical treatment and insulation of various environments. Its main feature is acoustic absorption, yielding the acoustical absorption coefficient over frequency. One of the ways to extract this parameter experimentally is with impedance tube (or Kundt's Tube) measurement via the transfer function method. However, the compression of the sample or the existence of gaps between the sample and the tube causes of uncertainty in the experiment. This research focuses upon the study of these effects. In order to analyze the influence of the boundary condition in the measurements, they were performed into two impedance tubes of slightly different internal diameters (both have the maximum frequency of analysis up to 6.4 kHz, approximately). Two different porous materials with samples of the same thickness were characterized using both tubes. In this situation, the procedure forces the influence of lateral leakage or compression of the samples against the tube to arise. Thus, this allows the analysis on the effect of the material structure in the acoustical absorption coefficient.

9:30

3aEAa6. Structural vibration damping of a vacuum-assisted toilet. Michael T. Rose, Joshua F. Kilts (Phys., Brigham Young Univ., Brigham Young University, Provo, UT 84602, rose.michael@byu.edu), B. Dagan Pielstick (Mech. Eng., Brigham Young Univ., Provo, UT), Scott D. Sommerfeldt, Kent L. Gee (Phys., Brigham Young Univ., Provo, UT), and Scott L. Thomson (Mech. Eng., Brigham Young Univ., Provo, UT)

Vacuum-assisted toilet noise can be unsettling and even uncomfortable. One common way to reduce noise levels is to damp structural vibrations that radiate sound. We investigated whether constrained layer damping (CLD) treatments could reduce the radiated noise level on a vacuum-assisted toilet. To find the structural anti-node locations of a toilet, we excited an airplane toilet with a shaker and scanned the velocity response of the inside of the bowl with a 3-dimensional scanning laser Doppler vibrometer (3D SLDV). We also scanned the bowl with an accelerometer during a repeated flush cycle. A microphone placed one meter above the bowl measured the radiated sound level. We applied 3M 4014, 3M 2552, Pyrotek Decidamp CLD, and Velcro individually to the bowl and determined the reduction in structural vibrations and sound radiation. The bowl's rim on the front vibrated the most. Structural vibrational energy concentrated around 100-400 Hz while radiated sound concentrated around 400 Hz–2 kHz. Applying damping materials reduced structural vibrations, sometimes by 20 dB. We conclude that CLD treatments are able to reduce structural vibrations. Further results of the investigation will be shown and discussed.

WEDNESDAY MORNING, 9 MAY 2018 GREENWAY D, 10:15 A.M. TO 11:45 A.M.

Session 3aEAb

Engineering Acoustics: General Studies on Transducers

Kenneth M. Walsh, Chair
K&M Engineering Ltd., 51 Bayberry Lane, Middletown, RI 02842

Contributed Papers

10:15

3aEAb1. Real-time source localization using phase and amplitude gradient estimator for acoustic intensity. Jacey G. Young (Dept. of Phys., St. Norbert College, 100 Grant St., Ste. 2324, De Pere, WI 54115, jacey.young@snc.edu), Joseph S. Lawrence, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Recent demonstrations of real-time source localization systems have used a variety of direction-finding methods. One such system uses a multi-microphone probe to estimate acoustic intensity [Inoue *et al.*, Proc. Meetings Acoust. **29**, 025001 (2016)], although the bandwidth of intensity estimates are traditionally limited by the microphone spacing. The Phase and Amplitude Gradient Estimator (PAGE) method [Thomas *et al.*, J. Acoust. Soc. Am. **137**, 3366–3376 (2015)] greatly extends the bandwidth of active intensity, providing an accurate method of source localization for a wide bandwidth. An initial system implemented in LabVIEW estimates active intensity using the PAGE method to identify the direction of the source location in real time. Using a paired webcam, the program then overlays the direction as a three-dimensional arrow onto a webcam image. A demonstration will be provided. Initial quantitative results show the accuracy of the system and limitations of the device. [Sponsored by the NSF REU program.]

10:30

3aEAb2. Focusing airborne acoustic vortex beams by means of active diffraction gratings. Ruben D. Muelas-Hurtado, Joao L. Ealo (School of Mech. Eng., Universidad del Valle, Calle 13 No100-00, Cali, Valle del Cauca 76001, Colombia, ruben.muelas@correounivalle.edu.co), Jhon F. Pazos-Ospina (Escuela Militar de Aviación - EMAVI, Colombian Air Force, Cali, Colombia), and Karen Volke-Sepulveda (Instituto de Física, Universidad Nacional Autónoma de México, Mexico City, Mexico D.F., Mexico)

New technologies to generate acoustic vortex beams (VB) have received increasing attention due to their interesting capabilities to transfer angular momentum to matter and to attract particles to the propagation axis. One of the recently reported methods to generate acoustic VB is the fabrication of active diffraction gratings, i.e., acoustic sources whose geometry and vibration mode emulate the field obtained after passing a plane wave through a given passive grating [1]. In this work, we extend this approach by using an active spiral zone plate to produce a beam with a helical and focused structure. This is obtained by gluing an electroactive ferroelectret film on top of a lower electrode structured on a printed circuit board, which in turn results easy and reliable. The broadband feature of the film makes it possible to generate VB throughout the range of interest for most applications in air. Experimental results are compared with numerical simulations. The proposed ultrasonic source opens up new possibilities for particle manipulation and rotation control of small objects. [1] R. D. Muelas H, J. F. Pazos-Ospina, and J. L. Ealo, "Active diffracting gratings for vortex beam generation in air," J. Acoust. Soc. Am. **141**(5), 3569–3569, 2017.

10:45

3aEAb3. Calibration and acoustic performance of sparse transducer array. Lane P. Miller, Stephen C. Thompson (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, lpm17@psu.edu), and Andrew Dittberner (GN Hearing, Glenview, IL)

In a previous research, a theoretical approach has been taken to predict, form, and optimize acoustic beam patterns from a sparse array of transducers. The current objective is to compare these theoretical results to experimentally obtained results from an assembled sparse array of transmit transducers. The array design and calibration procedure is presented. Acoustic beam patterns of the assembly are measured and compared to the theoretical model. Limitations of the array's performance are evaluated.

11:00

3aEAb4. Study on multi-channel speakers for controlling sound field of flat panel display. SungTae Lee (LG Display, Paju, South Korea), Hyung Woo Park, Myungjin Bae (IT, SoongSil Univ., 1212 Hyungham Eng. Bldg. 369 Snagdo-Ro, Dongjak-Gu, Seoul, Seoul 06978, South Korea, pphw@ssu.ac.kr), and Kwanho Park (LG Display, Paju, Korea (the Republic of))

Technological improvement in display industry, the organic light-emitting diode(OLED) panel manufacturer changed the quality of picture of the

screen from simple high definition to an augmented-reality or a virtual-reality. And OLED that construct with the actual pixels and emit light by themselves are becoming widespread, making the thickness thinner and the bezel thinner or even disappearing. However, these technology improvement made it difficult to achieve sufficient performance in a small space for sound quality, which is less important component of TV. In this study, we have developed a multi-channel flat panel speaker which is able to synthesis the sound field. That flat panel speaker consist with direct drive actuator and diaphragm as an outer—glass layer of OLED. This direct drive actuator speakers are used to realize multi-channel sound from a single OLED panel. Multi-channel speakers can control the sound field to set the direction of sound. This feature is a good condition for implementing mixed-reality. In previous research, it was confirmed that the viewer 's sense of reality increased when the position of sight and hearing were matched. Furthermore, we have been developed to increase the mixed reality, to improve the acoustic performance by adding directionality to the sound generated by the panel.

11:15

3aEAb5. A study on the sound generation at digital signage using OLED panel. Hyung Woo Park, Myungjin Bae (IT, SoongSil Univ., 1212 Hyungham Eng. Bulding 369 Snagdo-Ro, Dongjak-Gu, Seoul, Seoul 06978, South Korea, pphw@ssu.ac.kr), SungTae Lee, and Kwanho Park (LG Display, Paju, Korea (the Republic of))

With the development of ICT technology and convergence of media and ICT, smart media is spreading to people. The advertising industry through this smart screen is a rapidly growing future industry in the world. A new paradigm of bidirectional/customized features advertisement has become provided through IP-based new media such as smart TVs, portable smart devices, the Internet, IPTV (VOD), and digital signage. As one of the smart advertising industries, the digital signage is attracting attention as the fourth screen after TV, PC, and mobile with indoor/outdoor advertisement using information display. Advertising using the digital signage is installed in a lot of floating population place such as a subway station, a bus stop, an elevator, etc. Digital signage is developed with the display technology and the convergence of IT technology, but the development of appearance and screen configuration is mainly made. There are characteristics of high efficiency when communicate information with visual and auditory. However, up until now, digital signage has been difficult to provide vision and sound to people at the same time. The sound must be played through the screen; nevertheless, the smart screen does not pass the sound well, because of hardware. In this study, we introduce the technology that can generate direct sound using the characteristics of organic light-emitting diode (OLED) panel and introduce digital signage that can transmit video and sound simultaneously by using direct sound generation.

11:30

3aEAb6. A new lumped parameter model for the design of the free-flooded ring transducer. Kyounghun Been, Seungwon Nam (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), PIRO 416, Hyoja-dong, Nam-gu, Pohang-si, Gyeongbuk 790-784, South Korea, khbeen@postech.ac.kr), Haksue Lee, Hee-seon Seo (Agency for Defense Development, Changwon, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), Pohang, Kyungbuk, South Korea)

The free-flooded ring (FFR) transducer is the well-known low-frequency sound sources in underwater because of its broad operating frequency bandwidth and relatively small size. Many previous studies have been performed that predict the characteristics of an FFR transducer using an equivalent circuit model (ECM), a type of lumped parameter model (LPM) because an ECM is widely used to understand the properties of such transducers in the design process. However, it is quite difficult to predict the characteristics of an FFR transducer because the acoustic field is generated from its top and bottom openings, connected by the inner fluid, as well as the cylindrical ring surface. In this study, the authors investigated an ECM of an FFR transducer. The ECM consists of three parts: the piezoelectric ring, the cylindrical cavity, and the radiation load. In addition, an LPM which can consider mutual radiation loads was proposed to improve the accuracy of the model. The proposed models were compared and verified using commercial finite element method (COMSOL Multiphysics). We confirmed that LPM can predict characteristics of FFR transducer more accurately than ECM. [This work was supported by the National Research Foundation of Korea (NRF) grant funded by the Korea government (MSIP) (Grant No. 2016R1E1A2A02945515).]

Session 3aEDa

Education in Acoustics: Developing and/or Using Interactive Simulations for Teaching Acoustics

Andrew A. Piacsek, Cochair
Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926

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

Invited Papers

8:00

3aEDa1. Use of interactive simulations in pre-class learning activities. Tracianne B. Neilsen (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu)

To facilitate an effective in-class discussion, students must engage with the material prior to class. Assigned textbook reading is the traditional way students are asked to prepare, with possibly a reading quiz due before class. While carefully designed reading quizzes can be effective, a different approach is to use unscripted pre-class learning activities that provide hands-on interaction with the material. I created pre-class learning activities for the "Descriptive Acoustics" class at Brigham young University that use interactive simulations available on the internet. Provided with basic instructions, the students explore the simulation and write a description of their experience using terminology from the corresponding textbook chapter. Exploration followed by writing, even if incorrect, has increased the level of student understanding and provides a natural way to have students participate in class as they describe their experiences. The pre-class exposure to the interactive simulations also increases the efficacy of using the simulations during class, as each student is familiar with the interface and thus can better follow the in-class demonstration. Interactive simulations are a powerful active-learning tool for the current generation of students but only if they have had personal experience with the simulation prior to seeing it in class.

8:20

3aEDa2. Using interactive simulations to build understanding and promote scientific inquiry. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, andy.piacsek@cwu.edu)

Computer simulations of physical phenomena, with graphical output, have long been used to enable visualizations of processes that are inherently invisible, such as sound wave propagation, or that operate on time scales that make direct observation difficult. For students struggling to make sense of an abstract physical concept, such as the relationship between the propagation of a mechanical wave and the associated motion of the medium, an animated simulation of wave motion can accelerate the "aha!" moment of understanding. But to solidify and build on that understanding, students need to be able to formulate and test predictions. This can be accomplished with an *interactive* simulation, in which the user can adjust certain parameters of the model and view the response in the solution immediately. In this way, students can conduct virtual experiments. This presentation will use three examples of interactive simulations to demonstrate how interactive simulations can be used in a structured way to support specific learning objectives related to acoustics, and how they can be used to foster independent learning and scientific inquiry. Two examples come from a general education course in musical acoustics; the third example has been used in an advanced undergraduate/first-year graduate course in acoustics.

8:40

3aEDa3. Interactive (adjustable) plots and animations as teaching and learning tools. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, drussell@engr.psu.edu)

Interactive plots, such as are made possible with the Manipulate[ ] command in *Mathematica,* can be useful as teaching tools in the classroom, as well as learning aids for students outside of class. The use of adjustable sliders to change parameter values for an equation or system of equations allows for quick visual exploration of the effects of those changes on the resulting plot. Sometimes this visualization can help students understand difficult conceptual meaning hidden in mathematical expressions. Similarly, an adjustable interactive animation can effectively facilitate an understanding of concepts that are sometimes difficult to grasp from words or equations, or even "fixed" animations. This talk will demonstrate the creation of interactive plots and animations, and showcase several examples which the author has developed and used for teaching acoustics at the graduate and undergraduate levels. In addition, we will also discuss ways that students can be encouraged to create their own interactive plots or animations to aid in their own understanding.

9:00

3aEDa4. Simulations in acoustics and vibration education: What is the appropriate ratio of "learning to make" and "learning to use" for undergraduate students? Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

Within most undergraduate engineering curricula in the US, there is room for perhaps one or two acoustics and vibrations-related courses, which are generally offered as technical electives rather than as required classes. Numerical computation is also taught in one or two (typically required) courses. Yet, in the current environment, very powerful commercial simulation packages are available that can be used to solve complex real-world problems. Since there is no room in the current curriculum to teach both aspects fully, this begs the question, "What is the most appropriate mix of teaching?" Should one focus on the fundamentals, such that students learn the basics of various solution methods, but suffer limited exposure to the solution of real-world applications, or should one focus on the use of specific software packages that can solve real-world problems? These issues are explored and various examples are presented in hopes to further discussion of this topic in the community.

9:20

3aEDa5. Using animations to better understand acoustic impedance. Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., MS D446, Provo, UT 84602, bea@byu.edu)

It is very common for students to struggle to grasp a conceptual understanding of acoustic radiation impedance and input impedance. Often graduate students need exposure to impedance concepts in more than one course before they start to understand its importance and meaning. Instructors commonly suggest that impedance can be thought of as a resistance, ignoring the imaginary part of the impedance (the reactance), and the word "sloshing" is sometimes used to describe what the radiation reactance represents. This presentation will show some animations that have been developed in MATLAB to help students visualize the motion of the air particles near a vibrating surface. These animations utilize the relevant physical equations to observe the ratio of the pressure to the particle velocity. While these animations are not directly interactive, the user can change things like the frequency, source size, and excitation amplitude and re-create a new animation. Thinking of the impedance as the ratio of a potential quantity to a flow quantity can often be helpful, with this ratio having an in phase component and a component where the two quantities are 90 degree phase shifted.

9:40

3aEDa6. VSLM—The virtual sound level meter. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov)

Inexpensive sound level meters and low cost or free apps for smart phones and tables have made it much easier to introduce the concept of sound levels to a wide audience at very low cost. However, there are times when it is preferable to be able to take more complicated measurements that are generally only available on more expensive sound level meter. With fairly low cost digital recorders, one can now take a calibrated recording of an acoustic event and analyze the recording later. In this presentation, a free software package called VSLM: The Virtual Sound Level Meter is described. VSLM allows users to calibrate a recording and do many types of analysis including mimicking a sound level meter on fast, slow, Leq, octave, and 1/3 octave band analysis using fast FFT or digital meters that meet ANSI standards, high resolution spectral analysis, and create spectrograms.

10:00 – 10:20

10:20

3aEDa7. Numerical simulation and use of the PICUP framework for use in an undergraduate acoustics course. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

The use of numerical simulations in an undergraduate course introductory acoustics course can be a way to provide students with a method of exploring concepts that are difficult or costly to replicate in a classroom. For a course taught that serves a diverse group of students studying a large variety of majors, the importance of having multiple ways to present and revisit concepts should not be underestimated. Several simulations have been developed using Glowscript for use in a general education acoustics course. Additionally, the PICUP framework for development of exercise sets has been used to structure classroom activities in a way that encourages scaffolding of concepts from the most basic ideas to more complex applications. Examples of the Glowscript simulations and the development of PICUP exercises will be demonstrated.

10:40

3aEDa8. An interactive simulation of the filtered-x least-mean-squares algorithm for active noise control. Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

As part of a graduate acoustics course at Brigham Young University, students learn about principles of active noise control (ANC). One of the concepts covered is how a basic ANC algorithm functions. This presentation describes an interactive simulation of the filtered-x least-mean-squares (FXLMS) algorithm implemented in MATLAB. Available user-selected inputs include sampling frequency, type of signal to be controlled, number of filter coefficients, convergence coefficient, and system model parameters. As outputs, the students are able to observe the different signals, the attenuation, and the filter coefficients. Altering inputs and observing the simulation results motivates classroom discussion and facilitates students' gaining a working understanding of an ANC algorithm.

Contributed Paper

11:00

3aEDa9. Combining interactive simulation with scaffolded visualization in ESAIL to teach fundamental concepts of acoustical oceanography.
Jason E. Summers (ARiA, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com), Mark A. Ross, Zachary Walker, Paul Longerbeam, and Daniel Redmond (ARiA, Culpeper, VA)

To understand and interpret active-sonar returns and effectively employ active sonar systems for purposes such as antisubmarine warfare (ASW), operators must understand fundamental concepts of acoustical oceanography that govern how sonar interacts with the environment. Experts achieve high levels of performance in analysis and employment not by solving acoustic-propagation equations, but rather by employing simplified mental models that characterize the functional relationships governing how the environment affects acoustic propagation. Thus, a key goal in educating operators is helping them to develop and use such mental models. Here, we describe the development and employment of ESAIL, the Environment for Surface ASW Interactive Learning, a learning platform that combines interactive simulation with scaffolded visualization to facilitate instructors and students in conducting and assessing the outcome of "what if…?" experiments. In ESAIL, users are scaffolded in performing "what if…?" experimentation by tools that automatically classify user-modified acoustical environments as well as predictive acoustical models. This capability for direct exploration of cause-and-effect relationships with scaffolding to help develop expectations facilitates operators actively learning the mental models and mental-simulation capabilities experts use to interpret displays and reason about the underlying tactical scenario to achieve more optimal employment. [Work supported by the Naval Sea Systems Command.]

WEDNESDAY MORNING, 9 MAY 2018 — MIRAGE, 9:30 A.M. TO 12:30 A.M.

Session 3aEDb

Education in Acoustics: Hands-On for Middle- and High-School Students

Peggy B. Nelson, Cochair
Univ. of Minnesota, Minneapolis, MN 55455

Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, NY 11787

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 Minneapolis 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: Acoustics of Choirs and Vocal Ensembles

James P. Cottingham, Chair
Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, IA 52402

Chair's Introduction—9:00

Invited Papers

9:05

3aMU1. Traditional polyphony: Multipart singing in world cultures. Paul A. Wheeler (ECE, Utah State Univ., 1595 N. 1600 E., Logan, UT 84341-2915, pawheeler21@gmail.com)

Vocal polyphony has been part of the western musical culture since the 11th century. There are other cultures in the world, however, that use multipart singing in their traditional music that is not based upon the European classical style with parallel thirds and triad chords. This paper will discuss the use of vocal polyphony in several well-established traditions in the world, such as the Aka Pygmies from Central Africa, *Canto a tenore* from Sardinia, and iso-polyphony of Albania. It will contrast polyphonic types like parallel, drone, canonic, ostinato, and heterophonic polyphony. It will also illustrate how sound quality differs from the European classical style. An understanding of polyphonic traditions can assist us in understanding people whose cultures differ from our own.

9:25

3aMU2. The relation between choir size and choir dynamics. Ingo R. Titze and Lynn M. Maxfield (National Ctr. for Voice and Speech, Univ. of Utah, 136 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@utah.edu)

If six distinct levels of choir dynamics (*pp p mp mf f ff*) are to be achieved over two octaves of fundamental frequency in a choir section, how distinct are these levels? The just noticeable difference for sound level in a free field environment is 1-2 dB, while a doubling of loudness requires a 10 phon increase (10 dB SL increase at 1000 Hz). Most singers cannot double their vocal loudness 5 times. Generally the dynamic levels are 3–6 dB apart, depending on the individual *voice range profiles* of the singers. Overall choir size has no effect on dynamic range, unless the size is varied dynamically by not all singers singing all the time. A few loud voices dominate *ff* if everyone sings. To achieve an effective choir *pp,* only a few voices who can sustain very soft notes should sing. The dynamic range can be significantly limited when choral blend for loudness is imposed on a non-homogeneous group of singers.

9:45

3aMU3. Choir spacing vs choir formation: Long-term average spectra comparisons. James F. Daugherty (School of Music, Univ. of Kansas, 1530 Naismith Dr., Ste 448, Lawrence, KS 66045, jdaugher@ku.edu)

Choral methods books have long advised that repositioning choir singers solely according to voice part sung (i.e., choir formation) alters choral sound quality. To date, acoustical studies of choir formation have found no significant mean differences in long-term average spectra (LTAS) attributable to this strategy. Other investigations have indicated that changing the spatial distance between choir singers (i.e., choir spacing) yields significant LTAS differences. As yet, however, no study has compared the LTAS of choir spacing and choir formation conditions using the same choir. This paper reports experiments comparing LTAS of performances acquired from two microphone positions (conductor position, audience position) by three choirs of varied voicing (TTBB, SSAA, and SATB) singing in contrasting inter-singer spacing conditions (close, lateral, circumambient, and uneven) and voice-part formations (block sectional, mixed). Although recording venues and sung literature varied, each of the three choirs exhibited a pattern of significant mean timbral differences according to singer spacing conditions, but not according to formation conditions. Results were discussed in terms of the logics informing the spacing and formation approaches to modifying choir sound and implications for choral practice.

10:05

3aMU4. Effects of a straw phonation protocol on acoustic and perceptual measures of choirs. Jeremy N. Manternach (College of Education/School of Music, The Univ. of Iowa, 311 Communications Ctr., Iowa City, IA 52242, jeremy-manternach@uiowa.edu)

Voice instructors, choral directors, and voice professionals have long utilized semi-occluded vocal tract (SOVT) exercises to evoke efficient voicing from their students or clients. These exercises, which involve narrowing and/or lengthening the vocal tract as a means of increasing pharyngeal pressure, can include nasal consonants, lip or tongue trills, raspberries, or vocalizing through a tube or straw. Researchers have noted that the increased pressure reduces the amount of breath pressure required to initiate voicing (i.e., phonation threshold pressure) while maintaining or increasing acoustic output (e.g., sound pressure level, formant intensity). The result may be increased "vocal economy" or "vocal efficiency" (i.e., increased vocal output with decreased effort). Until recently, however, this

research was limited to effects on individuals. This paper includes a series of studies in which choirs of varied voicing (SATB, SSAA, and TTBB) sang unaccompanied pieces prior to and after performing voicing exercises through a small stirring straw. Long-term average spectra have indicated that most choirs maintained or increased acoustic output in the conglomerate, choral sound after the protocols. Singers have also perceived improved choral sound and decreased singing effort. These results may interest choral directors who wish to improve the vocal economy of their ensembles.

10:25

3aMU5. The barbershop sound: The characteristic timbre of the male barbershop quartet. Colin Drown (Phys. Dept., Truman State Univ., Kirksville, MO 63501, cdd5168@truman.edu) and James P. Cottingham (Phys., Coe College, Cedar Rapids, IA)

A feature of the homophonic barbershop style is an emphasis on major and dominant seventh chords which are sung with little vibrato and with pitches adjusted to achieve tuned chords in just intonation. A principal resource for the investigation reported here was the library of online recordings of the 2016 and 2017 international competitions sponsored by the Barbershop Harmony Society. These recordings provide a wealth of examples of the barbershop sound. They include hundreds of quartet samples performed in a context that features a uniform expectation of what constitutes good sound, along with uniform recording conditions and ranking by judges who serve as expert listeners. A large number of spectra were obtained for a loud, sustained major chord on an open vowel. Tuning of these chords was investigated, and for each sample a spectral contour was constructed with amplitudes normalized to simplify comparisons. From the consistency of these spectral contours it is possible to obtain a picture of the timbral features that distinguish the barbershop sound. In addition, patterns of relatively small differences in spectral envelope among the samples are strongly correlated with the ranking of the quartets by the competition judges.

WEDNESDAY MORNING, 9 MAY 2018 GREENWAY J, 9:00 A.M. TO 11:30 A.M.

Session 3aPA

Physical Acoustics: General Topics in Physical Acoustics I

Aaron Gunderson, Chair
The University of Texas at Austin, Austin, TX 78712

3a WED. AM

Contributed Papers

9:00

3aPA1. Development of a dual temporal-spatial chirp method for the generation of broadband surface acoustic waves. Marc Duquennoy, Dame Fall, Mohammadi Ouaftouh, Farouk Benmeddour, Salah-eddine Hebaz, Nikolay Samgin (Univ. Valenciennes, CNRS, Univ. Lille, YNCREA, Centrale Lille, UMR 8520 - IEMN, DOAE, F-59313, IEMN-DOAE, Campus Mont Houy, Valenciennes 59300, France, Salah-Eddine.Hebaz@univ-valenciennes.fr), Bogdan PIWAKOWSKI (Univ. Valenciennes, CNRS, Univ. Lille, YNCREA, Centrale Lille, UMR 8520 - IEMN, DOAE, F-59313, Villeneuve-d'Ascq, France), and Frederic JENOT (Univ. Valenciennes, CNRS, Univ. Lille, YNCREA, Centrale Lille, UMR 8520 - IEMN, DOAE, F-59313, Valenciennes, France)

This study deals with the optimization of transducers for Rayleigh-type Surface Acoustic Waves (SAW) generation. These transducers are based on Interdigital Transducers (IDT) and are specifically developed to characterize properties of thin layers, coatings, and functional surfaces. In order to characterize these coatings and structures, it is necessary to work with wide bandwidths IDT operating in the high frequencies. Therefore, in this study a spatial chirp-based on an IDT are realized for wideband SAW generation. The use of impulse temporal excitation (Dirac-type negative pulse) leads to a wide band emitter excitation but with significantly limited SAW output amplitudes due to the piezoelectric crystal breakdown voltage. This limitation can be circumvented by applying a temporal chirp excitation corresponding in terms of frequency band and duration to the spatial chirp transducer configuration. This dual temporal-spatial chirp method was studied in the 20 to 125 MHz frequency range and allowed to obtain higher SAW displacement amplitudes with an excitation voltage lower than that of the impulse excitation.

9:15

3aPA2. Acoustic vortex beam created with a continuous phase ramp lens and resulting radiation torque response of a sphere. Auberry R. Fortuner, Timothy D. Daniel, and Philip L. Marston (Phys. and Astronomy Dept., WSU, Washington State Univ., Pullman, WA 99164-2814, auberry.fortuner@wsu.edu)

An acoustic vortex beam was created in water using a lens with a phase ramp placed in front of a focused transducer. The lens was made from a polystyrene disk and CAD machine milled to have a continuous ramp in height about the center axis, with the ramp step height corresponding to a 2π phase difference between waves propagating in water and polystyrene at 500 kHz. This causes the transmitted beam through the lens to exit as a vortex beam with an angular phase ramp. It has been predicted [L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **136**, 2917–2921 (2014)] that the torque exerted by a vortex beam on an impenetrable sphere in a viscous fluid is proportional to the incident intensity due to dissipation of angular momentum in the viscous boundary layer near the sphere. This should result in steady rotation of the sphere with viscous drag proportional to angular velocity. Our upward-directed vortex beam was used to trap and spin a Styrofoam (closed foam) ball floating at the water surface. The spin rate was approximately proportional to the square of the source voltage as expected. [Work supported by ONR.]

9:30

3aPA3. Experimental measurement and analysis of acoustic wave propagation in water-filled steel pipeline using the iterative quadratic maximum likelihood algorithm. Zhao Li, Liwen Jing, Wenjie Wang, Yue Li, Amartansh Dubey (Electron. and Comput. Eng., Hong Kong Univ. of Sci. and Technol., Rm 2427, Academic Bldg., Clear Water Bay, Kowloon, Hong Kong, Hong Kong 999077, Hong Kong, eelizhao@ust.hk), Pedro Lee (Dept. of Civil and Natural Resources Eng., Univ. of Canterbury, Christchurch, New Zealand), and Ross Murch (Electron. and Comput. Eng., Hong Kong Univ. of Sci. and Technol., Hong Kong, Hong Kong)

Acoustic wave propagation (up to 50 kHz) within a water-filled steel pipeline is studied using laboratory experiments. The experiments were carried out in a 6 m length of cylindrical stainless steel pipeline using acoustic transducers to acquire signals from 100 locations uniformly spaced along the longitudinal axis of the pipe. By applying the iterative quadratic maximum likelihood algorithm (IQML) to the experimental results, parameters such as wavenumbers, attenuations, and mode amplitudes were accurately extracted for individual modes from the measurement data. We found that the IQML algorithm could extract these parameters more accurately in situations where the measurement data had low signal to noise ratio as compared to other algorithms such as Prony's method. A very good match was obtained between the experimental results and predictions from an analytical waveguide model for the wavenumber dispersion curves, attenuations, and acoustic power characteristics of the axisymmetric and non-axisymmetric modes. Additional physical explanations of the propagation phenomena in the pipeline waveguide were obtained using the experimental results and analytical model.

9:45

3aPA4. Qualitative analysis of mode transitions in bottle-shaped resonators with waterfall plots. Bonnie Andersen, David Eldred, and Josh Dimond (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84057, bonniem@uvu.edu)

A closed bottle-shaped resonator consists of the coupled neck and cavity. Such a system yields avoided crossings where the resonance of the neck matches that of the cavity when one bottle dimension is varied. Self-sustained oscillations within the bottle are generated thermoacoustically. Mode transitions were previously observed to occur at the same position within a few millimeters when a piston controlled with a translation stage was moved with a manual control to adjust the cavity length. The dominant mode was recorded using a power spectrum of the signal measured with a pressure sensor. In this study, the piston motion is automated and eight neck/cavity combinations were tested at three different piston speeds and at various input powers. The input powers were adjusted to just above thermoacoustic onset and not to exceed thermal limits of the materials used. Waterfall plots allow the visualization of the time evolution of the power spectrum where intensity is plotted both as a function of time and frequency. Qualitatively, the transitions occur at the same place within the cavity after a threshold input power is reached. Interestingly, overtones appear in most cases to be harmonics of the fundamental with either all or only odd harmonics present.

10:00

3aPA5. Determining the mass of gas in large collection volumes with acoustic and microwave resonances. Keith A. Gillis, Jodie G. Pope, Michael Moldover (Sensor Sci. Div., National Inst. of Standards and Technol., 100 Bureau Dr., Mailstop 8360, Gaithersburg, MD 20899-8360, keith.gillis@nist.gov), and James B. Mehl (Sensor Sci. Div., National Inst. of Standards and Technol., Orcas, Washington)

We characterized a 1.8 m^3, quasi-spherical resonator, a pressure vessel informally known as the "big blue ball" or "BBB." The BBB will be the collection volume of a NIST-designed gas-flow standard that will operate at pressures up to 7 MPa. Using microwave resonance frequencies, we determined the volume of the BBB filled with argon as a function of pressure (up to 7 MPa) and temperature (near ambient). The measured pressure- and temperature-dependences of the BBB's volume are consistent with the published properties of carbon steels. We filled the BBB with 220 kg of argon in weighed increments of approximately 20 kg. We also determined the mass m_{acoust} of argon in the BBB by measuring the pressure, the acoustic resonance frequencies f_{acoust} of 3 modes, and using the known thermophysical properties of argon. The values of m_{acoust} were within $\pm$0.03 % of the gravimetrically determined masses. In a typical application, the BBB will never be isothermal. Nevertheless, the acoustic resonance frequencies quickly and accurately determine the average temperature and mass of the gas in the BBB despite significant, long-lived thermal gradients that are created by pressure changes. A theoretical model for effect of temperature gradients on the acoustic resonance frequencies is presented.

10:15–10:30 Break

10:30

3aPA6. Acoustical and optical resonances of two-sphere systems. Cleon E. Dean (Phys., Georgia Southern Univ., PO Box 8031, Math/Phys. Bldg, Statesboro, GA 30461-8031, cdean@georgiasouthern.edu)

Two sphere systems show strong morphology dependent resonances even for two spheres of identical composition [G. W. Kattawar and C. E. Dean, Opt. Letts. 8, 48–50 (1983).]. How do analogous fluid spheres respond in the analogous situations? Do they show similar or different resonances? Preliminary investigations suggested that the response of fluid bispheres do not show as rich a panoply of resonances [C. E. Dean and J. P. Braselton, JASA, **141**, 3735 (2017).]. How much do these systems differ? Would alternate boundary conditions restore some lost resonances? This talk attempts to answer these and other questions through the use of theoretical computational acoustics models.

10:45

3aPA7. Inferring atmospheric surface-layer properties from wind noise in the nocturnal boundary layer. Carl R. Hart (U.S. Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, carl.r.hart@usace.army.mil) and Gregory W Lyons (U.S. Engineer Res. and Development Ctr., Champaign, IL)

Uncertainties in the state of the atmosphere, to a large extent, limit the prediction accuracy of outdoor sound propagation. In particular, event sound propagation requires accurate knowledge of wind speed and temperature profiles, spatially averaged over the path of propagation. In a stable quasi-steady nocturnal boundary layer, wind speed and temperature gradients follow a scaling that is asymptotically independent of altitude and depends on Monin-Obukhov similarity parameters. Since these parameters describe the near-surface profiles of wind speed and turbulent intensity, which in turn are known to govern wind noise, it is anticipated that a connection exists between Monin-Obukhov parameters and the statistics of wind noise. It is expected that these parameters may be inferred from wind noise sensed by screened microphones. Ambient noise data collected in the southwest U.S. are analyzed for the purpose of examining whether Monin-Obukov similarity parameters may be inferred from wind noise. We explore the consequences of establishing inferences with a priori distributions for the similarity parameters, and utilizing wind noise data from microphones at one or more altitudes.

11:00

3aPA8. Absorption and dispersion of low-frequency sound waves in the main cloud layers of Venus. Adam Trahan and Andi Petculescu (Univ. of Louisiana at Lafayette, University of Louisiana at Lafayette, 240 Hebrard Blvd, Rm 103, Lafayette, LA, ajt6261@louisiana.edu)

We present predictions for the acoustic wavenumbers at low frequencies in the condensational cloud layers of Venus, occurring between 50 and 70 km, approximately. While the general thermodynamics of Earth clouds is well understood, that of Venusian clouds is still a matter of debate. Venus' clouds are formed primarily of H_2O and H_2SO_4 vapors and aqueous sulfuric acid droplets, the fluxes of which are not fully constrained due to the few *in situ* observations. This study is based on and adapted from the terrestrial model of Baudoin *et al.* (J. Acoust. Soc. Am. **130**, 1142 (2011)). Inside the clouds, the Navier-Stokes equations of continuum fluid mechanics are used for the gaseous (dry + vapor) and liquid phases of H_2O and H_2SO_4, combined with the equations describing the evaporation/condensation processes; the gaseous phase is treated as an ideal gas. Outside the clouds, in the CO_2—dominated atmosphere, the van der Waals equation of state is used.

[The work was supported by a Grant from the Louisiana Space Consortium (LaSPACE).]

11:15

3aPA9. Effect of clustered bubbly liquids on linear-wave propagation.
Yuzhe Fan, Haisen Li, Chao Xu, and Baowei Chen (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Nangang District, Harbin, Heilongjiang 150001, China, fanyuzhe@hrbeu.edu.cn)

The spatial distribution of bubble liquids strongly affect linear-wave propagation in bubbly liquids. The classical understanding suggests that using positional correlations of the bubbles, such as "hole correction" or Perkus-Yevick approximation, describes the spatial information of bubble clouds. However, bubbly liquids are observed experimentally with complex bubble ensembles that take the form of clusters, filaments, and clouds which cannot be intuitively described by these pair-correlation functions and need to be better clarified. To achieve this purpose, a three-dimensional random model, the Neyman-Scott point process, is proposed to describe bubbly liquids with clustering. Base on this method, we study the influence of such phenomenon on acoustic dispersion and attenuation relations. A formula for effective wavenumber in bubbly liquids is derived, based on self-consistent method. Comparing with the equation of Commander and Prosperetti [J. Acoust. Soc. Am. **85**, 732 (1989)], our results show that the clustering can suppress peaks in the attenuation and the phase velocity as functions of the frequency. Further, we provide a numerical simulated method. A clustered bubbly liquid is simulated with strict mathematical method and the statistical informations are obtained through unbiased statistical approach. Through the results, we quantificationally analyze the influence of estimated value on predictions.

WEDNESDAY MORNING, 9 MAY 2018 — NICOLLET D2, 7:55 A.M. TO 10:55 A.M.

Session 3aPPa

Psychological and Physiological Acoustics: Physiology Meets Perception

Sarah Verhulst, Cochair
Dept. Information Technology, Ghent Univ., Technologiepark 15, Zwijnaarde 9052, Belgium

Antje Ihlefeld, Cochair
Biomedical Engineering, New Jersey Institute of Technology, 323 Martin Luther King Blvd, Fenster Hall, Room 645, Newark, NJ 07102

Invited Paper

7:55

3aPPa1. The eardrums move when the eyes move: A multisensory effect on the mechanics of hearing.
Kurtis G. Gruters (Psych. and Neurosci., Duke Univ., Fayetteville, NC), David L. Murphy, Cole D. Jenson (Psych. and Neurosci., Duke Univ., Durham, NC), David W. Smith (Psych., Univ. of Florida, Gainesville, FL), Christopher Shera (Caruso Dept. of Otolaryngol., Univ. of Southern California, Los Angeles, CA), and Jennifer M. Groh (Psych. and Neurosci., Duke Univ., LSRC Rm B203, Durham, NC 27708, jmgroh@duke.edu)

Interactions between sensory pathways such as the visual and auditory systems are known to occur in the brain, but where they *first* occur is uncertain. Here, we show a novel multimodal interaction evident at the eardrum. Ear canal microphone measurements in humans ($n=19$ ears in 16 subjects) and monkeys ($n=5$ ears in 3 subjects) performing a saccadic eye movement task to visual targets indicated that the eardrum moves in conjunction with the eye movement. The eardrum motion was oscillatory and began as early as 10 ms before saccade onset in humans or with saccade onset in monkeys. These eardrum movements, which we dub Eye Movement Related Eardrum Oscillations (EMREOs), occurred in the absence of a sound stimulus. The EMREOs' amplitude and phase depended on the direction and horizontal amplitude of the saccade. They lasted throughout the saccade and well into subsequent periods of steady fixation. We discuss the possibility that the mechanisms underlying EMREOs create eye movement-related binaural cues that may aid the brain in evaluating the relationship between visual and auditory stimulus locations as the eyes move.

Contributed Paper

8:15

3aPPa2. A model of the neuronal processing of interaural time disparities. Jörg Encke (Bio-Inspired Information Processing, Tech. Univ. of Munich, Boltzmannstrasse 11, Garching 85748, Germany, joerg.encke@tum.de), Florian Völk (WindAcoust., WinDC, Germany), and Werner Hemmert (Bio-Inspired Information Processing, Tech. Univ. of Munich, Garching, Germany)

The auditory system of humans and other mammals is able to use interaural time and intensity differences as well as spectral cues to localize sound sources. One important cue for localizing low-frequency sound sources in the horizontal plane are interaural time differences (ITDs), which are first analyzed in the medial superior olive (MSO) in the brainstem. Results from electrophysiological and psychoacoustic studies suggest ITD encoding in the relative activities of neuronal populations in the two brain hemispheres. This contribution first explores the neuronal encoding of fine-structure ITDs using a physiologically motivated spiking neuronal-network model of the mammalian MSO. Results from this model confirm robust ITD encoding in the relative activity of the MSOs in both hemispheres. Based on the neuronal-network simulations, a simple probabilistic model of subsequent ITD processing is proposed. This simplified model connects the neuronal responses of the two hemispheres with different hearing sensation properties, such as lateral position, spatial extent, or number of spatially separable sensations. First predictions from the model are discussed with regard to the results of a series of psychoacoustic experiments on the lateralization of pure-tone impulses presented via headphones.

Invited Papers

8:30

3aPPa3. Evidence for the origin of the binaural interaction components of the auditory brainstem response. Sandra Tolnai and Georg Klump (Animal Physiol. & Behaviour Group, Dept. for Neurosci., Carl von Ossietzky Univ. Oldenburg, Carl-von-Ossietzky-Str. 9-11, Oldenburg 26129, Germany, sandra.tolnai@uni-oldenburg.de)

The binaural interaction component (BIC) is discussed as a potential tool to objectively measure listeners' binaural auditory processing abilities. It is obtained from auditory brainstem responses (ABRs) by subtracting the sum of the ABRs to monaural left and monaural right stimulation from the ABR recorded under binaural stimulation. The sources of the BIC, however, have not yet been agreed upon. Candidate source regions are the lateral and medial superior olives (LSO and MSO, respectively) in the superior olivary complex where excitatory and inhibitory inputs converge. Our study aims at identifying the source of the BIC. Simultaneously to ABRs, we recorded local-field potentials (LFPs) and single-unit (SU) responses from the LSO and MSO of ketamine/xylazine-anaesthetised Mongolian gerbils and derived LFP-related and SU-related BICs the same way as ABR-related BIC. We then compared the properties of LFP-related and SU-related BICs with the ABR-related BICs. LFP-related BICs recorded in the LSO did not mirror the characteristics of the ABR-related BIC while the SU-related BIC did. In the MSO, neither LFP-related nor SU-related BIC mirrored ABR-related BICs. This suggests that the output of LSO units but not MSO units contribute substantially to the generation of the ABR-related BIC.

8:50

3aPPa4. Neural bases of complex sound perception: Insights from an avian speech mimic. Kenneth S. Henry (Otolaryngol., Univ. of Rochester, 601 Elmwood Ave., Box 629, Rochester, NY 14642, kenneth_henry@urmc.rochester.edu) and Laurel H. Carney (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Many complex sounds including speech contain periodic fluctuations in signal level known as amplitude modulation (AM). Auditory-nerve fibers encode AM primarily through response synchrony to signal modulation. In contrast, neurons of the midbrain encode AM through both response synchrony and substantial changes in average discharge rate. Specifically, many midbrain neurons with band-pass modulation tuning show enhanced discharge activity in response to a limited range of modulation frequencies. Here, we describe several recent studies on the relative importance of envelope synchrony vs. average rate coding in the midbrain for AM perception. Behavioral and neurophysiological experiments were conducted in the budgerigar, an avian species with the unusual capacity to mimic speech. Budgerigars were found to exhibit human-like behavioral sensitivity to a variety of complex sounds including AM stimuli and synthetic vowels with triangular and natural spectral envelopes. Most neurons in the budgerigar midbrain showed band-pass modulation tuning. Whereas either response synchrony or average discharge rate was sufficient to account for behavioral performance in some cases, only response synchrony could explain behavioral thresholds for (1) detection of low modulation frequencies and (2) formant-frequency discrimination in noise. These results highlight the significance of midbrain envelope synchrony for perception of complex sounds. [R00-DC013792 and R01-DC001641.]

9:10

3aPPa5. Is there a relationship between perceptual difficulties and auditory functioning at the periphery and the brainstem in children suspected with auditory processing disorder? Sriram Boothalingam (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, 1975 Willow Dr., Goodnight Hall, Rm 482, Madison, WI 53706, boothalingam@wisc.edu), Sangamanatha Veeranna (National Ctr. for Audiol., Western Univ., London, ON, Canada), Chris Allan (Commun. Sci. and Disord., Western Univ., London, ON, Canada), David Purcell, and Prudence Allen (National Ctr. for Audiol., Western Univ., London, ON, Canada)

The test battery typically used for the diagnosis of auditory processing disorder (APD) is highly heterogeneous, with an emphasis on the central auditory nervous system. As such, the peripheral auditory system is typically only screened for the presence of an overt hearing loss. Our previous work suggested that children suspected of APD (sAPD) have atypically sharp cochlear tuning when measured using stimulus frequency otoacoustic emission (SFOAE) group delay. In the present work, we extend our previous findings and test the hypothesis that cochlear tuning influences auditory brainstem response (ABR) latencies. Our hypothesis is based on filter theory, which suggests that a sharper filter will take longer to build-up and ring longer. We predicted that sharper cochlear filters in sAPD should result

in delayed ABR wave latencies and will be associated with poorer performance on speech perception tests. Preliminary data from 16 sAPD and 6 typically developing children show a positive correlation between cochlear tuning and ABR peak I latency. Cochlear tuning explains a significant proportion of the variance in ABR peak I latency ($R^2 = 0.25$). Further behavioral results and implications for inclusion of auditory peripheral examination in APD tests will be discussed.

9:30–9:45 Break

9:45

3aPPa6. Transient developmental hearing loss leads to disruption of striatal function in adults. Todd M. Mowery (CNS, New York Univ., 4 Washington Pl., 1008, New York, NY 10003, tm106@nyu.edu)

Transient hearing loss during development can lead to deficits in the auditory perceptual abilities of children. These deficits arise from both peripheral ear and central neural dysfunction along the auditory neuraxis. Even if hearing loss resolves or is treated, persistent learning deficits and language impairments can occur. This suggests that regions downstream of the primary auditory neuraxis can be disrupted by brief auditory deprivation. One such region, which is highly involved in language development, is the auditory striatum. Thus I asked how the development of cellular properties in the auditory striatum are affected by hearing loss, and how these changes affect the learning of an auditory task. I used the gerbil and a brain slice preparation to assess how the expression of long term potentiation (LTP), which is a biomarker of learning, correlated with behavioral performance in adult animals that had undergone developmental hearing loss. I found that the cellular expression of LTP corresponded with the rapid acquisition of the behavioral task, and that developmental hearing loss delayed both the expression of LTP and the correlated behavioral acquisition. This suggests that striatal dysfunction could play a role in language acquisition impairments in children with hearing loss.

10:05

3aPPa7. Understanding pitch perception through physiological, modeling, and behavioural methods. Kerry M. Walker (Physiol., Anatomy & Genetics, Univ. of Oxford, Sherrington Bldg., Parks Rd., Oxford OX1 3PT, United Kingdom, kerry.walker@dpag.ox.ac.uk)

Pitch is a salient perceptual quality that underlies our musical experience, interpretation of speech, and ability to attend to one of multiple speakers. Yet the brain mechanisms that support this key percept remain unclear. We have measured the pitch discrimination of humans, ferrets, and mice on a 2-alternative forced choice task. These experiments show that species differ in their weighting of harmonic and temporal envelope cues for pitch judgments. By applying existing computational models, we demonstrate how cochlear filter properties can explain some of these species differences. We have also used microelectrode recordings in behaving ferrets to understand how individual cortical neurons represent pitch. We find that: (1) cortical activity correlates better with ferrets' pitch judgements than stimulus f0; and (2) inhibition can shape neural representations of f0 when animals are actively engaged in a pitch discrimination task, compared to when they are passively listening to the same sounds. We are now using 2-photon calcium imaging, during which we can measure the precise spatial position and responses of large numbers of individual neurons simultaneously, to better understand how and where pitch is extracted within the cortical microcircuit.

Contributed Papers

10:25

3aPPa8. Dynamic allocation of listening effort when listening to speech. Matthew Winn (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St, Seattle, WA 98105, mwinn83@gmail.com)

Measuring listening effort is an essential part of understanding the communication ability of a person with hearing loss. In this study, we are using pupillometry to measure how listener effort changes over the course of an utterance, in cases where the listener has reason to allocate effort earlier or later in time. Listeners completed a sentence-recognition task where high- or low-context sentences were followed by various stimuli (noise/digit sequences/speech) or silence. In cases where listeners can "restore" misperceived words with the aid of context, we see significant late-occurring reductions in pupil size when a sentence is followed by silence. This alleviation of cognitive activity is shown to be systematically disrupted by sounds (noise or speech) that are played after the end of the sentence. Furthermore, we show evidence of reallocation of effort earlier in time in cases where that disruption is expected. This underscores a largely underappreciated challenge in auditory perception, which is not limited to the coding stimulus itself, but also planning and post-stimulus processing. This phenomenon might underlie individuals' difficulty in real-life scenarios where a single mistake cannot be corrected in time, and cascades forward to disrupt perception of later utterances.

10:40

3aPPa9. Using functional near-infrared spectroscopy to assess auditory attention. Min Zhang and Antje Ihlefeld (Biomedical Eng., New Jersey Inst. of Technol., 323 Martin Luther King Blvd, Fenster Hall, Rm. 645, Newark, NJ 07102, ihlefeld@njit.edu)

Speech intelligibility in background sound can improve when listeners attend to the target. Using functional magnetic resonance imaging, previous work demonstrates that auditory attention alters Blood Oxygenation Level Dependent (BOLD) signals in the lateral frontal cortex (LFCx; Michalka *et al.*, Neuron 2015), at least when directed to spatial cues. Using functional near-infrared spectroscopy (fNIRS), we recorded BOLD responses from LFCx in 54 normal-hearing participants, each performing one of three auditory experiments. The weights of general linear modeling (GLM) fits of the fNIRS recordings were compared using analysis of variance. In experiment 1, when listeners selectively attended to stimuli differing in pitch and space, both left and right LFCx showed significant activation ($p = 0.01$ and 0.02 for left versus right LFCx). In contrast, passive listening to speech-shaped noise did not cause significant changes in LFCx activation ($p = 0.3$ and p 0.1 for left versus right LFCx). Experiment 2 confirmed these results, revealing comparable fNIRS traces when attending to pitch alone, versus attending to space and pitch. Experiment 3 further strengthened these results by controlling for eye movements. Result suggests that fNIRS is a viable tool for assessing whether or not a listener deploys auditory attention to perform a task.

Session 3aPPb

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

William Yost, Chair
Arizona State University, PO Box 870102, Tempe, AZ 85287

Chair's Introduction—11:00

Invited Paper

11:05

3aPPb1. Cortical mechanisms underlying perception in complex auditory scenes. Shihab Shamma (Elec. and Comput. Eng. & Inst. for Systems Res., Univ. of Maryland College Park, AV WIlliams Bldg, College Park, MD 20742, sas@umd.edu)

Humans and many animals can easily attend to one of multiple similar sounds, segregate it, and follow it selectively over time even in extremely noisy environments. I shall review the psychological and neural underpinnings of this perceptual feat, explaining how it fundamentally depends on the representation of sound in the auditory cortex. I shall then outline how the attributes of simultaneous sounds (pitch, timbre, and location) are disentangled into separate streams, and bound into unified sources, by the temporal coherence of the cortical responses they evoke and their rapid adaptive properties. Recent neurophysiological results in support of these ideas will be discussed, especially in animals that have been trained to segregate and attend to the speech stream of a target speaker in a mixture. Finally, these findings will be related to analogous tasks in other sensory systems (visual object segregation in crowded scenes), leading to biologically inspired computational algorithms that may perform these tasks with no prior information or training.

Session 3aSC

Speech Communication: Session in Memory of James J. Jenkins

Kanae Nishi, Cochair
Boys Town National Research Hospital, 555 N. 30th Street, Omaha, NE 68131

Terry L. Gottfried, Cochair
Psychology, Lawrence University, 711 E. Boldt Way, Appleton, WI 54911

Linda Polka, Cochair
School of Communication Sciences & Disorders, McGill University, 2001 McGill College Avenue, 8th floor, SCSD, Montreal, QC H3Z 1Z4, Canada

Chair's Introduction—8:40

Invited Papers

8:45

3aSC1. Jim Jenkins—Mentor extraordinaire. Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Box 357920, Seattle, WA 98195, pkkuhl@u.washington.edu)

James J. Jenkins (J^3 to his students) was the quintessential mentor. As a graduate student at the University of Minnesota in the late 1960s in Speech Communication, Jim was a member of my advisory committee, and a strong advocate of my interests and pursuits in the area of language and the brain. Jim set up a research fellowship for me with Dr. Hildred Schuell, a clinical scientist he deeply respected and collaborated with on publications, who was Director of the Aphasia Section in the Neurology Service at the VA Hospital in Minneapolis. I began to study the effects of cerebrovascular accident (stroke) on language, and was thrilled to be there. Unfortunately, Dr. Schuell succumbed after a sudden onset of cancer 9 months after my fellowship began. I remember asking Jim, "now what?" J^3's reply, and the 3-month assignment he gave me, changed my life, setting me on a course that I remain on to this day. I was never able to thank J^3 enough for his sage advice. Jim's talent at mentoring students—his ability to understand where they might shine—is something that inspires me every day.

9:05

3aSC2. When infants encounter infant speech. Linda Polka (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., Montreal, QC H3Z 1Z4, Canada, linda.polka@mcgill.ca)

Over the past four decades, we have learned a great deal about how infants perceive and decode the speech spoken around them and directed to them. Yet we know very little about how infants perceive their own vocalizations or speech with the unique vocal properties of an infant talker. This leaves a serious gap in our understanding of infant language development. In this talk, I will present findings from a new line of research that begins to address this neglected aspect of infant speech development by exploring how infants perceive speech with infant vocal properties. The findings suggest that access to infant speech has a broad and significant impact—influencing receptive, expressive, and motivational aspects of speech development. I will highlight some of the valuable lessons I learned under the mentorship of James J. Jenkins that have steered me along this rich and exciting line of research.

9:25

3aSC3. Recent findings on children's lexical tone development: Implications for models of speech development. Puisan Wong (Speech and Hearing Sci., The Univ. of Hong Kong, Rm. 757 Meng Wah Complex, Pokfulam NA, Hong Kong, pswResearch@gmail.com)

Lexical tone is important for a majority of the world's languages. Prior research consistently reported that children mastered lexical tone production between 1;6 to 2;6 of age, supporting the prevailing notion of theories of speech acquisition that children acquire suprasegmental features early and rapidly before mastering the full set of segmental features. However, intriguingly, studies that examined children's Cantonese tone perception found that children failed to accurately identify Cantonese tones in monosyllabic words until after eight years old, five years after children's full mastery of tone production at three years old, which questioned early acquisition of suprasegmental features and the widely accepted assumption in speech acquisition models that speech perception precedes speech production. **Wong, Schwartz, and Jenkins (2005)** started a series of studies using a new paradigm that controlled lexical expectation in tone rating to examine lexical tone development in children and reported strikingly different results. Monosyllabic and disyllabic tones produced by five and six years old Mandarin-speaking children growing up in the U.S. and in Taiwan and Cantonese-speaking children growing up in Hong Kong were not adult-like. Tone production development was slower than tone perception development. The findings shed new light on universal models and theories of speech acquisition.

9:45

3aSC4. Voice quality: Speaker identification across age, gender, and ethnicity. Sonja Trent-Brown (Psych. Dept., Hope College, 35 E 12th St., Holland, MI 49423, trentbrown@hope.edu)

Listeners can perceptually identify speaker gender and ethnicity at better than chance guessing for adult speakers (Lass *et al*., 1978; Thomas & Reaser, 2004; Trent-Brown *et al*., 2012). These findings are supported across varying levels of phonetic complexity and across temporal manipulations that impact perceptual processing. There is less evidence regarding the extent to which these patterns hold for child speakers and whether listener gender/ethnic background and experience influence accuracy of identification. African American and European American male and female adult speakers were recorded producing passage-, sentence-, and word-length (/hVd/) stimuli including 11 General American English vowels. African American and European American male and female child speakers, ages 8–12, were recorded producing sentence and word stimuli. A temporal manipulation introduced to distort perceptual expectations presented stimuli to listeners in both forward and reversed conditions. Listeners were male and female African American and European American undergraduate students. Significant outcomes were observed for accuracy across speaker gender, speaker ethnicity, phonetic complexity, and temporal condition. In addition to accuracy of identification, listener confidence ratings, identification reaction times, and confidence reaction times also showed differential outcomes with respect to speaker age, gender, and ethnicity.

10:05–10:25 Break

10:25

3aSC5. "If a perceptual problem exists, what are its roots?" Types of cross-language similarity, hypercorrection, and unexplored root canals. Ocke-Schwen Bohn, Anne A. Ellegaard, and Camila L. Garibaldi (Dept. of English, Aarhus Univ., Aarhus DK-8000, Denmark, engosb@hum.au.dk)

James Jenkins had a keen interest in the nature of perceptual problems of nonnative listeners and in the perceived similarity of speech sounds. This presentation reports on a series of experiments which examined how well four different types of similarity predict nonnative speech perception. The different types are ecphoric and perceptual cross-language similarity, perceived within-language similarity, and acoustic similarity. One set of experiments examined how well cross-language perceptual assimilation (English to Danish) and within- language similarity ratings (English-English) of English consonants by Danish listeners predict Danes' identification of English consonants. Another set of experiments explored the roots of English listeners' discrimination problems for the closely spaced Danish unrounded front vowels by relating these problems to two types of perceived cross-language similarity (ecphoric and perceptual) and to acoustic similarity. Results suggest that each of the four types of similarity accounts for some of the perceptual problems, but none does so exhaustively, probably because the root system of these problems is affected by additional factors including nonnative listeners' hypercorrection and by perceptual asymmetries. The clear conclusion from these experiments is that Jim's question will keep us busy for quite some time. [Work supported by Carlsberg Foundation and *Inge Lehmanns Legat af 1983*.]

10:45

3aSC6. Tracking the time course of individuals' perception and production of coarticulated speech. Patrice S. Beddor (Dept. of Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109, beddor@umich.edu)

Pioneering research by James J. Jenkins and colleagues has demonstrated the importance for speech perception of the dynamic information specified across the syllable. Collaborative work in our lab builds on this tradition of studying listeners' use of dynamic information by investigating perception of coarticulatory information as it unfolds in real time. We are currently studying listener-specific strategies for attending to coarticulatory dynamics, and are asking whether these strategies are linked to that individual's own production patterns. I will report the results of experiments that test the hypothesis that individuals who attend to coarticulatory information especially closely in perception also produce more consistent and extensive coarticulation. The perceptual measure is the time course of participants' use of coarticulated vowel nasality in CVNC words as measured via eye tracking; the production measure is the time course of these participants' nasal airflow while producing CVNC words. Results support our hypothesis: participants who produced earlier onset of coarticulatory nasalization were, as listeners, more efficient users of that information as it became available to them. Thus, a listener's use of the dynamics of speech is predicted, to some degree, by that individual's production patterns. [Work supported by NSF grant BCS-1348150.]

11:05

3aSC7. Catching a rabbit with a tetrahedron: A contextualist approach. Valeriy Shafiro (Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu)

A defining characteristic of Jim Jenkins' long and productive research career was his ability to reconcile contradictory experimental evidence within a coherent conceptual framework that leads to novel insights and experimental approaches. Grounded in the deep knowledge of history of psychology, Jim's involvement and expertise in different aspects of human behavior, including language, speech, memory, attention, visual, and auditory perception, resulted in important theoretical breakthroughs (and countless stimulating discussions). One such work is the tetrahedral model of memory experiments. It considers four large clusters of experimental variables: subjects, materials, scoring tasks, and testing procedures, whose mutual interactions determine experimental outcomes. [J.J. Jenkins "Four points to remember: A tetrahedral model of memory experiments," in *Levels of Processing in Human Memory*, ed. LS Cermak, FIM Craik, (1979)]. Initially proposed to account for inconsistent findings of memory experiments, the model's applications have expanded to other areas of perceptual training and education as a guide to maximize learning outcomes. Following Jim's lead, I will illustrate how the tetrahedral model can (1) improve the understanding of research findings from two ecologically significant sound classes, speech and environmental sounds, in normal hearing and cochlear implant listeners, and (2) guide the design of effective auditory training programs.

11:25–11:55 Panel Discussion

Session 3aSP

Signal Processing in Acoustics and Underwater Acoustics: Co-Prime Arrays and Other Sparse Arrays I

R. Lee Culver, Cochair
ARL, Penn State University, PO Box 30, State College, PA 16804

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

Invited Papers

9:00

3aSP1. Sampling with semi-coprime arrays. Kaushallya Adhikari (Louisiana Tech Univ., 600 Dan Reneau Dr, Ruston, LA 71270, adhikari@latech.edu)

This research introduces a new array geometry called Semi-Coprime Array (SCA) that has the potential to significantly increase the degrees of freedom of a Uniform Linear Array (ULA). An SCA interleaves three ULAs (Subarray 1, Subarray 2, and Subarray 3). Each SCA has two underlying coprime integers (M and N). Subarray 1 and Subarray 2 have undersampling factors based on the coprime integers M and N, while Subarray 3 is a full ULA with the number of sensors determined by the undersampling factors of Subarray 1 and Subarray 2. Interleaving the three subarrays results in a highly sparse non-uniform linear array. Taking the minimum of the three subarrays' outputs produces a pattern that is devoid of aliasing, yet offers the resolution of a full ULA with equal aperture. An SCA offers closed form expressions for sensor locations which many existing sparse arrays do not. Compared to other existing sparse arrays that offer closed form expressions for sensor locations, coprime arrays and nested arrays, an SCA saves more sensors and matches the peak side lobe height of a full ULA with less extension.

9:20

3aSP2. Investigations on n-tuple coprime arrays. Dane R. Bush and Ning Xiang (Architectural Acoust., Rensselaer Polytechnic Inst., 2609 15th St, Troy, NY 12180, danebush@gmail.com)

Coprime arrays so far combine two sparsely-spaced subarrays, undersampled by factors of M and N, in order to achieve MN degrees of freedom. To ensure that the grating lobes of each subarray can be largely eliminated, M and N must be coprime. In number theory, sets of pairwise coprime numbers can exceed just two numbers. The current work extends the theory to include coprime linear arrays with an arbitrary number, n, of subarrays. A triple coprime array comprised of $n = 3$ equal-aperture subarrays with M, N, and O elements, undersampled by factors of NO, MO, and MN, respectively, may use just $M + N + O - 1$ shifts to observe MNO directions. The design frequency of such an array not only exceeds the Nyquist spatial sampling limit, but is also greater than that of a standard (double) coprime array with equivalent aperture and number of elements. A triple coprime array is constructed with subarrays of $M = 3$, $N = 4$, and $O = 5$ and measured in a simulated free field condition. Experimental validation of the array confirms that the triple coprime array can also observe lower frequencies up to the design frequency. This paper discusses advantages and practical significance of n-tuple coprime microphone arrays over conventional double coprime linear arrays.

9:40

3aSP3. Effects of lag redundancy on array beam patterns and redundancy patterns for extended co-prime arrays. Andrew T. Pyzdek (Graduate Program in Acoust., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu) and David C. Swanson (Appl. Res. Lab., Penn State Univ., State College, PA)

Lag redundancy describes the number of times a given spatial lag is measured by an array. For sparse arrays, the reduction of lag redundancies is generally desirable for lowering cost and processing overhead. However, as lag redundancies determine the natural weighting function of an array's spectral estimates, the pattern of redundancies dictates the natural beam pattern of a given array. We consider the impact of array sparsity on sidelobe level and array gain in both weighted and unweighted conventional beamforming. This analysis is then applied to co-prime arrays, for which redundancy patterns can be determined based solely on the choice of array length and co-prime factors. It is also shown that the spatial positions of lag redundancies are semi-regular, a desirable feature for sampling some complex environments.

10:00

3aSP4. Time difference of arrival-based acoustic source localization and separation technique for sparse arrays. Mingsian R. Bai (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

An acoustic source localization and separation technique intended for sparse arrays is suggested in this paper. Unlike conventional beamforming approaches, the proposed technique is based on time difference of arrival (TDOA) information. This effectively eliminates the spatial aliasing problem due to overly large inter-element spacing of sensors. This method begins with estimation of the TDOAs between at least 4 sensors for 3-dimensional sound fields, with the aid of the generalized crosscorrelation-phase transformation (GCC-PHAT) algorithm and the subspace-based time delay estimation algorithm. Next, a constrained least squares (CLS) algorithm is applied to locate the source based on the above-estimated TDOAs. Once the source location is found, the source signal is extracted by using the Tikhonov regularization (TIKR) algorithm and minimum variance distortionless response (MVDR) beamformer. Regularization parameter is selected with a special search procedure to best preserve the audio quality in the inverse solution process. Experiments and listening tests are undertaken to validate the proposed TDOA-based localization and separation technique.

10:20–10:35 Break

10:35

3aSP5. Applying concepts and tools from signal processing on graphs (SPG) to problems in array signal processing. Eldridge Alcantara, Les Atlas (Elec. Eng., Univ. of Washington, Seattle, Box 354090, Seattle, WA 98195, eealcant@uw.edu), and Shima Abadi (Mech. Eng., Univ. of Washington, Bothell, Bothell, WA)

Signal processing on graphs (SPG) is an emerging area of research that extends well-established data analysis concepts and tools to support a special type of signal where data samples are defined on the vertices of a graph. Since SPG emerged in 2013, fundamental operations such as filtering, the Fourier transform, and modulation have been formally defined that uniquely consider and take advantage of the underlying complex and irregular relationship between data elements which is captured mathematically by a graph. The purpose of this study is to analyze the applicability of SPG to array signal processing. We show that signals defined on a graph, or graph signals for short, are natural models for data collected over a line array of sensors. We also apply existing SPG processing algorithms to array signal data and investigate and probe whether SPG can help increase array gain.

10:55

3aSP6. Frequency-wavenumber analysis with a sparse array. Yonghwa Choi, Donghyeon Kim, and Jea Soo kim (Korea Maritime and Ocean Univ., 727 Taejong-ro, Youngdo-Gu, Busan 49112, Busan KS012, South Korea, hwa1470@naver.com)

In underwater acoustics, there has been many studies for finding the target direction using beamforming technique. When receiving a signal with a frequency higher than the design frequency of the array, it is difficult to estimate the direction of the signal due to spatial aliasing. In this study, we propose a method of estimating the direction of the frequency signal higher than the design frequency of the array by using frequency-wavenumber analysis. When the frequency-wavenumber analysis is performed, the striation pattern appears, and it is confirmed that the slope of the striation remains constant even if the spatial aliasing occurs. The direction of the signal was estimated by visual inspection and it was verified with SAVEX15 data.

11:15

3aSP7. Frequency-difference wavenumber analysis with a sparse array. Donghyeon Kim (Dept. of Convergence Study on the Ocean Sci. and Technol., KIOST-KMOU Ocean Sci. and Technol. School, 727 Taejong-ro, Yeongdo-Gu, 253, Ocean Sci. and Technol., Korea Maritime and Ocean University, Busan ASIKRKS012BUSAN, South Korea, donghyeonkim@kmou.ac.kr), Yonghwa Choi, Seongil Cho (Korea Maritime and Ocean Univ., Busan, South Korea), gihoon byun (Dept. of Convergence Study on the Ocean Sci. and Technol., KIOST-KMOU Ocean Sci. and Technol. School, Busan, N/A, Korea (the Republic of)), and Jea Soo kim (Korea Maritime and Ocean Univ., Busan, South Korea)

Frequency (f)- wavenumber (k) analysis can be used to estimate the direction of arrival (DOA) [J. Acoust. Soc. Am. **69**, 732–737 (1980)]. When the receiver is a sparse array that is not suitable for conventional plane-wave beamforming, it adversely causes aliasing error due to spatial sampling, thus many striation patterns can emerge in f-k domain. In this study, we propose frequency-difference wavenumber analysis that is motivated frequency-difference beamforming [J. Acoust. Soc. Am. **132**, 3018–3029 (2012)]. It is found that this approach can mitigate (or eliminate) such aliasing effect, which extends its applicability to the robust DOA estimation. Numerical simulation and experimental results are presented, and a major drawback is discussed.

Session 3aUW

Underwater Acoustics: Target Scattering in Underwater Acoustics: Imaging, Spectral Domain, and Other Representations I

Daniel Plotnick, Cochair
Applied Physics Lab., Univ. of Washington, Seattle, WA 98105

Timothy Marston, Cochair
APL-UW, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—8:00

Contributed Paper

8:05

3aUW1. Numerical determination of Green's functions for far field scattering solutions. Aaron M. Gunderson and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78758, aaron.gunderson01@gmail.com)

Typical finite element target scattering results are evaluated over a small physical domain and then propagated to the far field using the Helmholtz-Kirchhoff integral. This process allows for direct far field data-model comparison and is substantially faster than evaluation over a large domain. It does, however, require knowledge of the Green's function of the target's environment. For underwater buried targets, the two-medium Green's function pertaining to the water and sediment is often not readily known and can be cumbersome to analytically solve or approximate, particularly when surface roughness or volume inhomogeneity is to be incorporated. Instead, the Green's function can be determined directly through numerical methods. A direct process for numerical Green's function determination in an arbitrary two-medium environment is proposed, and is used to determine the far field scattering from buried targets. Results are compared to experimental scattering records on buried elastic targets, and to other models where the Green's function was determined/approximated analytically. [Work supported by Applied Research Laboratories IR&D and ONR, Ocean Acoustics.]

Invited Paper

8:20

3aUW2. Quantitative ray methods for scattering by tilted cylinders. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

After a review of ray methods and phenomena associated with high frequency scattering by spheres and cylinders in water viewed broadside, generalizations to tilted cylinders relevant to spectral and imaging domains will be summarized. These extensions were found to be useful for meridional as well as helical ray backscattering enhancements associated with leaky (or supersonic) waves on shells [F. J. Blonigen and P. L. Marston, J. Acoust. Soc. Am. **112**, 528–536 (2002)]. For such enhancements, Fermat's principle is useful for identifying ray paths of interest. In the case of helical waves (and in the broadside special case) the scattering amplitude can be expressed in terms of a Fresnel patch area where the guided wave is excited on the shell. Fresnel patches also give insight into the relatively large magnitude of meridional ray contributions. The coupling coefficient is proportional to the radiation damping of the leaky wave. For some shells it is necessary to take into account the anisotropy of the phase velocity. Computational benchmarks include scattering into the meridional plane by tilted infinite cylinders. Related phenomena include time-frequency domain features and enhancements from waves crossing truncations and from subsonic and negative group velocity guided waves. [Research supported by ONR.]

Contributed Papers

8:35

3aUW3. Backscattering enhancement due to weakly damped, leaky, guided waves on cylinders and associated images. Timothy D. Daniel, Sterling M. Smith, and Philip L. Marston (Phys. and Astronomy Dept., WSU, Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu)

In this work, distinct but related targets were studied using a circular synthetic aperture sonar system. Backscattering data from a solid brass cylinder were recorded. The response of the target was significantly spread out in time because leaky guided waves were weakly radiation damped. They gave significant backscattering contributions for a range of tilt angles. The associated meridional and helical wave contributions are also obvious in the frequency domain. (For comparison helical contributions are relatively weak for aluminum cylinders.) The second target is a bi-metallic cylinder made of a 1:1 brass cylinder bonded to a 3:1 aluminum cylinder. For a single-material cylinder, only 90 degrees of data are necessary due to symmetry. This target breaks that symmetry and requires a full 180 degrees of rotation. The distinction between the brass and aluminum ends is discernible in both the time and frequency domain. The join between the two metals affects the timing of certain guided waves that no longer travel the whole length of the compound cylinder. Image reconstruction was performed using Fourier based algorithms for both targets. The location of the compound cylinder joint is deducible from the location of features in the image domain. [Research supported by ONR.]

8:50

3aUW4. Separation of multiple backscattered echoes using dictionary updating sparse method. Xiangxia Meng, Xiukun Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Harbin, Heilongjiang 150001, China, mengxiangxia@hrbeu.edu.cn), and Andreas Jakobsson (Lund Univ., Lund, Sweden)

Sparse signal representation method has been proven to be effective to estimate and separate the multiple backscattered echoes from underwater target. Usually, the transmitted signal is used as a template to generate the dictionary. It is efficient under assumptions that the knowledge of signal components is perfectly known, or the deviations from the dictionary can be ignored. However, the properties of the echoes differ from the clean reference signal because of the scattering mechanism. In contrast, it has necessity to extract these deviations to have deep insight of the target. Herein, we propose a sparse method that allows the dictionary to be updated. The initial dictionary is defined as a usual way, i.e., based on the transmitted signal. Then, we use a constrained optimization problem to generate dictionary that differs from the original one, introducing an updating procedure. We introduce two constraints, one is to penalize the updated dictionary to be sparse and the other one is to ensure the elements do not deviate too much from the original dictionary. In this way, the resulting dictionary can be used to reconstruct the backscattered echoes, which is illustrated to be efficient with high accuracy by simulated and experimental results.

9:05

3aUW5. Experimental study on characteristics of echoes reflected by a cylindrical object in underwater multi-path channel. Fangyong Wang, Shuanping Du, and Jiao Su (National Key Lab. of Sci. and Technol. on Sonar, Hangzhou Appl. Acoust. Res. Inst., No. 715 Pingfeng Rd., LiuXia St., Hangzhou, Zhejiang 310023, China, sklwfy@yahoo.com)

Research work on characteristics of echoes scattered by a cylindrical object located in underwater multi-path is introduced in the paper. A sound scattering experiment of a cylindrical object in water-filled tank is carried out, the purpose of which is to give answers to questions of how the multi-path channel corrupt the characteristics of echoes from an elastic body in underwater multi-path channel and how many target-related features can still be extracted in both time and frequency domain which can be used for classification or feature-based detection. Detailed data processing results are shown and analyzed, which may be useful to research on signal processing method of sonar target detection and classification.

9:20

3aUW6. Coherent fusion of multi-aspect scans of UXO targets from CLUTTEREX16. Timothy Marston (APL-UW, 1013 NE 40th St., Seattle, WA 98105, marston@apl.washington.edu)

During the ONR and SERDP sponsored 2016 CLUTTEREX experiments, a large number of UXO were placed in a target field. These UXO were scanned by a sonar system mounted on a linear rail. Following each scan, the UXO were rotated in place by increments of 20 degrees to build up a multi-aspect acoustic characterization of each target. An automated process was subsequently developed to coherently fuse the results of these rotations to form 180-degree wide acoustic characterizations that could be used to represent the target in the complex image, frequency vs. aspect, or time vs. aspect domains. This talk will discuss the fusion process and the results for UXO and clutter objects in the target field.

9:35–9:50 Break

Invited Papers

9:50

3aUW7. Use of time, space, frequency, and angle data products in understanding target-in-the-environment scattering physics. Kevin Williams, Steven G. Kargl, and Aubrey L. Espana (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, williams@apl.washington.edu)

Broadband, multi-aspect time domain data of scattering from targets placed on sediment interfaces contains a wealth of information about the target and the modifications to the target scattering due to its location in the ocean environment. Different data products derived from the data can be used together to gain a better insight into the overall physical processes at play. We use a 2:1 aluminum cylinder and a 2:1 aluminum pipe to demonstrate the utility of different types of data products. Domains examined include imaging in the horizontal plane, frequency response as a function of look angle, and time domain response as a function of look angle. We then examine these same domains using relatively narrow windows of time with varying start times. This analysis allows visualization of the evolution of the physical mechanisms involved. [Sponsored by the Office of Naval Research and the Strategic Environmental Research and Development Program.]

10:05

3aUW8. Representation trade-offs for the quality assessment of acoustic color signatures. J. D. Park (The Penn State Univ. Appl. Res. Lab., P.O.Box 30, State College, PA 16804, jdanielpark@psu.edu), Daniel Cook (Georgia Tech Res. Inst., Smyrna, GA), and Alan J. Hunter (Univ. of Bath, Bath, United Kingdom)

Acoustic color is a representation of the spectral response over aspect, typically in 2-D. This representation aims to characterize the structural acoustic phenomena associated with an object with frequency and aspect as the two axes of choice. However, the strongest acoustic color signatures are typically the geometric features of an object due to geometric scattering. Naturally, these geometric features are well-represented as straight-line signatures in the spatial spectrum representation with the axes horizontal and vertical wavenumbers. Elastic responses due to other scattering mechanisms such as Rayleigh scattering or Mie scattering are typically smaller in magnitude, and are not easily recognizable or separable from the stronger responses. This work explores variants of acoustic color, to find intuitive representations for these types of scattering responses, and to assess their signature quality.

10:20

3aUW9. Elastic features in time-frequency representations for target classification. Rodolfo Arrieta, David E. Malphurs, Iris Paustian (Naval Surface Warfare Ctr. Panama City Div., 110 Vernon Ave., Panama City, FL 32407, rodolfo.arrieta@navy.mil), and Raymond Lim (Naval Surface Warfare Ctr. Panama City Div., Panama City, FL, FL)

Time-frequency representations aid in the interpretation of backscattered chirps, allowing a physical basis for selection of features for target classification using machine learning approaches. However, trade-offs in resolution, computational efficiency, noise reduction, and cross term rejection are made in these 2-D representations. Since our goal is the isolation of physically-meaningful features that will aid in the classification of targets, we are mainly concerned with the ability of these representations to reject noise. Towards this end, we have developed synthetic signals designed to robustly enhance the energy in the elastic portion of the acoustic backscattered return; thus, increasing the SNR for difficult targets. Previously, we demonstrated a capability to class-separate a diverse group of targets using features derived from time-frequency representations of signals collected at high SNR target aspects. In recent work this capability is being extended to other target aspects where SNR is weak by using our synthetic signals. This is demonstrated with cylindrical targets, where physics-based target models helped select the most robust elastic modes to use in the design of these synthetic signals. Results based on data collected from underwater unexploded ordnance and simple cylindrical targets will be presented. [This work was sponsored by SERDP and NSWC PCD.]

10:35

3aUW10. Utilization of aspect angle information in synthetic aperture images. Daniel Plotnick (Univ. of Washington, Applied Physics Lab., Seattle, WA 98105, plotnd@apl.uw.edu) and Timothy Marston (Univ. of Washington, Seattle, WA)

Synthetic aperture sonar (SAS) involves the creation of high resolution images of a scene via scattered signals recorded at different locations. Each pixel of the reconstructed image includes information obtained from multiple aspects due to the changing position of the sources/receivers. The aspect dependent scattering at each pixel may be exploited to provide additional information about the scene; we will present a framework for converting and utilizing multi-aspect data, as well as several examples. This aspect dependency may be leveraged to separate objects of interest from the background, to understand the local bathymetry, or for visualizing acoustic shadowing in circular aperture sonar (CSAS) images. Additionally, the aspect dependence of low-frequency elastic scattering from objects may be used to understand underlying scattering physics; a laboratory example is presented.

Contributed Paper

10:50

3aUW11. Laboratory simulation of backscattering by a hemisphere in a boundary wave-guide with an external source. Auberry R. Fortuner and Philip L. Marston (Phys. and Astronomy Dept., WSU, Pullman, WA 99164-2814, auberry.fortuner@wsu.edu)

While there have been various investigations of backscattering when a source and a target are situated in the same wave-guide, the situation when the source is external to the wave-guide is less well explored. For example, a target buried in a mud layer over sand [K. L. Williams, J. Acoust. Soc. Am. 140, EL504 (2016)], the mud layer may act as a wave-guide. The target signatures when the source is in the water column above the mud are modified. A laboratory experiment has been performed to isolate wave-guide effects on target backscattering, using a simplified wave-guide in water. The wave-guide is defined between a free water surface and a thin partially transparent acrylic plate placed below the water surface. A soft spherical target is half-exposed at the free-surface and source-receiver is below the plate. A geometric method-of-images model gives the timing arrivals of wave-guide echoes from an infinite number of back-scattering paths that undergo multiple reflections in the wave-guide and gives the frequency response of the wave-guide by summation of the contributions. This model gives a closed-form result in the far-field limit, and is compared to the experimental results in the time and frequency domains. [Work supported by ONR.]

11:05

3aUW12. Synthetic aperture sonar speckle noise reduction performance evaluation. Marsal A. Bruna, David J. Pate, and Daniel Cook (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, marsal.bruna@gtri.gatech.edu)

SAS (Synthetic Aperture Sonar) imagery always contains speckle, which is often thought of as a kind of multiplicative noise with respect to the underlying scene reflectivity. Speckle arises from the coherent interference of waves backscattered by rough surfaces within a resolution cell. Numerous image processing algorithms have been proposed to reduce speckle while preserving image features. Frequently, these algorithms are evaluated using actual SAS imagery; where the underlying noise-free image is unknown. The lack of noise-free images along with a limited number of images can lead to an incomplete estimation of algorithm performance. The use of various image quality and speckle reduction metrics is also required to accurately assess performance. In this paper, a unified framework using both simulated and real imagery is used to analyze the performance of prominent algorithms, such as multilook and anisotropic diffusion methods.

11:20

3aUW13. Simulating an object response with synthetic aperture sonar inverse imaging. James Prater (NSWC PCD, 110 Vernon Ave., Panama City, FL 32407, james.l.prater@navy.mil)

Synthetic aperture sonar (SAS) algorithms were developed in order to produce high resolution imagery from wide-beam wideband data. The high data rates associated with SAS data have fostered the development and implementation of efficient beamforming algorithms, many of which were originally developed for synthetic aperture radar. Some of these beamforming algorithms are invertible, and inverse beamforming techniques have been developed for SAS data analysis, where imagery is inverted in order to produce aspect-dependent data. Since imagery is invertible, this process can also be used as a simulation to produce stave-level data from a simulated image. This paper describes a modeling approach where inverse imaging techniques are applied in order to generate simulated stave-level sonar data. Using this approach, an image is generated from assumed object geometry and reflectance and inverted in order to generate stave-level data. The stave level data is then beamformed specifically to produce frequency vs. aspect data products. Since the inverse imaging process assumes specular reflection, features present in the frequency vs. aspect data are solely due to the surface reflectance and object shape. Comparison of inverse image simulated data with physics based simulations or field-collected data are shown and the limitations of the comparison are discussed.

11:35

3aUW14. Characterization of internal waves in synthetic aperture sonar imagery via ray tracing. David J. Pate, Daniel Cook (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, david.pate@gtri.gatech.edu), Anthony P. Lyons (Univ. of New Hampshire, Durham, NH), and Roy E. Hansen (Norwegian Defence Res. Establishment, Kjeller, Norway)

Synthetic aperture sonar imagery often captures features that appear similar to sand waves but are actually pockets of denser water traveling as isolated waves along the seafloor. These pockets of cold water refract acoustic waves like a lens, causing intensity peaks and shadows that resemble medium to large scale sand waves. This work uses dynamic ray tracing to predict the intensity return as affected by refraction. First, we explore the nature of the intensity pattern created by internal waves of various shapes and sizes. Then, we use an optimization-based approach to solve the inverse problem: given an intensity pattern, determine the size, shape, and location of the internal wave that created it.

Session 3pAA

Architectural Acoustics: AIA CEU Course Presenters Training Session

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066

Chair's Introduction—1:00

Invited Papers

1:05

3pAA1. Architectural acoustics short course presentation material. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects, called "Architectural Acoustics." An architect can earn one continuing education unit (CEU) by attending this short course, if it is presented by a qualified member of TCAA. The course covers topics in sound isolation, mechanical system noise control, finish treatments, and implementation of quality acoustical spaces. This paper will cover the course material in order to prepare and qualify potential presenters. In order to qualify as an authorized presenter for this AIA/CES short course, attendance at this special session and membership in TCAA are required.

2:05

3pAA2. Architectural acoustics continuing education course—Presenter registration and reporting requirements. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

The Technical Committee on Architectural Acoustics (TCAA) is a Registered Provider in the American Institute of Architects (AIA) Continuing Education System (CES). The TCAA has developed a standardized introductory short course for architects called "Architectural Acoustics," for which attendees can earn one continuing education unit (CEU). This paper will cover the administrative requirements of the AIA/CES, to prepare potential presenters. These requirements include the proper handling of paperwork so that AIA members may receive credit for the course. The manner in which the course is given is also dictated by AIA requirements. TCAA membership and attendance at this workshop are required to qualify as an authorized presenter for this AIA/CES short course. Of course, anyone is free to register with the AIA to provide their own CEU program. However, the advantages of participating in this program are that the TCAA short course is already prepared, it is pre-approved by the AIA, and the registration fees are paid by the Acoustical Society of America.

Session 3pAB

Animal Bioacoustics: Lessons on Auditory "Perception" from Exploring Insect Hearing

Norman Lee, Chair
Biology, St. Olaf College, 1520 St. Olaf Avenue, Northfield, MN 55057

Chair's Introduction—1:10

Invited Papers

1:15

3pAB1. Insect ear biomechanics: Passive and active processes that can inspire acoustic sensors. James F. Windmill (Dept. of Electron. & Elec. Eng., Univ. of Strathclyde, 204 George St., Glasgow G1 1XW, United Kingdom, james.windmill@strath.ac.uk)

Biological systems provide an incredible wealth of archetypes that have emerged through evolutionary processes. Hearing organs are a good example of how different solutions, and adaptions, across different animal taxa, can often converge to solve similar sensory problems. Hearing has evolved independently multiple times across the insects, and the diversity of these biological solutions therefore provides a wealth of inspiration for the creation of novel acoustic sensors. Some biological solutions can be considered as purely passive mechanical constructs that accomplish some processing of the incoming sound. This talk will consider examples of this, including the frequency discrimination of the locust ear, and the directionality and wideband response of different moth ears. Several insects also display active hearing processes, whereby energy is used to actively change the hearing response, bearing some similarity to the processes found in the mammalian inner ear. This talk will thus discuss how some moths actively tune their ear, and how the mosquito ear actively utilizes gain and compression. Finally, the talk will discuss some examples of different acoustic sensors from the University of Strathclyde that take inspiration from these passive and active biological processes.

1:35

3pAB2. Stay tuned: Active processes tune tree cricket ears to maintain a match to temperature-dependent song frequency. Natasha Mhatre, Gerald Pollack, and Andrew Mason (Dept. of Biological Sci., Univ. of Toronto, 1265 Military Trail, Mason Lab, Scarborough, ON M1C 1A4, Canada, natasha.mhatre@gmail.com)

Tree cricket males produce tonal songs, which are used to attract mates and to interact with other males. In other crickets, the mechanical properties of their peripheral auditory system have evolved to have resonant modes at the communication frequency. The tree cricket auditory system, however, is not passively tuned to song frequency. Tree crickets exploit an active amplification process to tune hearing to song frequency. However, their song frequency increases with temperature, presenting a problem for tuned listeners. We show here that the actively amplified frequency increases with temperature, and shifts mechanical and neuronal auditory tuning to maintain a match with changing song frequency. We also find that in tree crickets active amplification does not provide greater sensitivity or dynamic range than that observed in other crickets that lack active amplification. Thus, the primary adaptive function of active amplification is to ensure that auditory tuning remains matched to conspecific song frequency, despite changing environmental conditions and signal characteristics.

1:55

3pAB3. A neural circuit for song pattern recognition in the cricket brain. Berthold Hedwig (Zoology, Univ. of Cambridge, Dept. of Zoology, Downing St., Cambridge CB2 3EJ, United Kingdom, bh202@cam.ac.uk), Stefan Schoeneich (Inst. for Biology, Univ. of Leipzig, Leipzig, Germany), and Konstantinos Kostarakos (Inst. of Zoology, Karl Franzens Univ., Graz, Austria)

Acoustic communication is based on amplitude and frequency modulation of sound signals. Temporal features of the signal require processing by central auditory neurons, the brain circuits, however that detect temporal features are poorly understood. We show how five neurons in the brain of female field crickets form an auditory feature-detector circuit for the pulse pattern of the male calling song, and exhibit properties of a delay-line and coincidence-detection mechanism. The network receives its direct input from a single ascending auditory interneuron. An internal delay that matches the pulse period of the calling song is established by a non-spiking brain neuron. In response to a sound pulse, it generates a transient inhibition that triggers a delayed rebound depolarization. The direct input and the delayed responses converge in a coincidence detector neuron, which responds best to the pulse pattern of the species-specific calling song as the rebound activation of the non-spiking neuron coincides with the response of the ascending interneuron to the subsequent sound pulse. The output of the coincidence detector neuron is further processed by a feature detector neuron to suppress unselective responses and background activity. The circuit reveals principal mechanisms of sensory processing underlying the perception of temporal auditory patterns.

2:15

3pAB4. Stimulus specific adaptation in the auditory system of insects: Can a single neuron learn? Johannes Schul and Ryan Yost (Biological Sci., Univ. of Missouri, 207 Tucker Hall, Columbia, MO 65211, schulj@missouri.edu)

Stimulus-specific adaptation (SSA) is the suppression of a neuron's activity to repetitive stimuli while maintaining responsiveness to infrequent signals. In *Neoconocephalus* katydids, one auditory interneuron (TN-1) shows strong SSA in oddball paradigms, when standard and oddball pulses differ in carrier frequency. SSA occurred for pulse rates from >140 Hz down to 1 Hz. At fast repetition rates (>100 Hz), responses to the common pulses ceased, while oddballs elicit single spikes. At slower rates (<50 Hz), both standard and oddball pulses elicited spiking, with responses to oddballs being significantly larger, comparable to SSA described in vertebrate hearing systems. At slow rates, SSA also occurred when the shape of standard and oddball pulses differed, while having identical spectral properties. We identified at least two dendritic mechanisms that contribute to SSA at fast pulse rates (>100 Hz). The mechanisms underlying SSA at slow pulse rates are less well understood; likely, dendritic Ca^{2+}-gated currents contribute to SSA at slow pulse rates. At slow rates, response reduction to repeated pulses resembles habituation as oddball pulses cause dishabituation, i.e., the response to the following standard pulse is larger than that to the preceding one. Whether this habituation is generated within TN-1 or by its synaptic inputs remains an open question.

2:35

3pAB5. Making high-stakes decisions in complex acoustic environments: Revisiting the Cocktail Party problem in a multimodal sensory context. Daniel R. Howard (Biological Sci., Univ. of New Hamsphire, UNH Spaulding Hall G32, 38 Academic Way, Durham, NH 03824, daniel.howard@unh.edu), Norman Lee (Biology, St. Olaf College, Northfield, MN), and Carrie L. Hall (Biological Sci., Univ. of New Hamsphire, Durham, NH)

Ambient noise in its many forms represents an ecological reality and evolutionary driver that influences numerous expressions of animal behavior, especially those associated with communication. Noise often extends across sensory modalities, resulting in channel-specific effects that can range from non-additive to synergistic. Here, we describe how noise modality influences female preference for a preferred signal trait (low dominant frequency) in the lek-mating prairie mole cricket, *Gryllotalpa major*, a species whose mating system has evolved in the ecological context of both biotic and abiotic noise. We conducted two-choice playback experiments with females presented with male signals of the preferred vs. non-preferred trait in the context of isomodal (male chorus), cross-modal (substrate-borne vibration) and multimodal noise (both) conditions. We found that female preference for lower DF signals was robust to isomodal noise, but preference was weakened in treatments with a cross-modal stimulus only. Female performance in multimodal noise was comparable to control and isomodal treatments, suggesting that while cues obtained via the subgenual organs can mask salient airborne information, biotic noise may act as a releaser from this interference.

Session 3pBAa

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

Kevin J. Haworth, Chair
University of Cincinnati, 231 Albert Sabin Way, CVC3940, 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 US $500 as first prize, US $300 as second prize, and US $200 as 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.

1pBA4. Wideband transskull refocusing of ultrasound beams using dual-mode ultrasound arrays: Ex vivo results
Student author: Hasan Aldiabat

1pBA7. Intranasal administration of temozolomide combined with focused ultrasound to enhance the survival of mice with glioma (A Pilot Study)
Student author: Dezhuang Ye

1pBA8. Broadband transskull multi-focus wavefront synthesis
Student author: Parker O'Brien

2aBA8.A nonlinear grid-search inversion for cortical bone thickness and ultrasonic velocities
Student author: Tho Tran

2pBA2. Estimating liver fat fraction from ultrasonic attenuation and backscatter coefficient measures in adults with nonalcoholic fatty liver disease
Student author: Lucia Albelda

2pBA11. Quantifying nonlinear elasticity modulus of tissue-like solids using acoustic radiation force
Student author: Danial Panahandeh-Shahraki

4pBAa1. In vivo ultrasound thermal ablation controlled using echo decorrelation imaging
Student author:Mohamed Abbass

4pBAa2. Patient-specific large-volume hyperthermia in the liver for ultrasound-enhanced drug delivery from thermosensitive carriers
Student author: Brian Chu

4pBAa6. Blood coagulation monitoring using acoustic levitation
Student author: Vahideh Ansari

Session 3pBAb

Biomedical Acoustics and Physical Acoustics: Induction Mechanisms for Bubble Nucleation II

Jeffrey B. Fowlkes, Cochair
Radiology, Univ. of Michigan, 3226C Medical Sciences Building I, 1301 Catherine Street, Ann Arbor, MI 48109-5667

Ronald Roy, Cochair
Engineering Science, University of Oxford, Parks Road, Oxford OX1 3PJ, United Kingdom

Invited Paper

1:40

3pBAb1. Tailoring cavitation nuclei for biomedical applications. Tyrone M. Porter (Boston Univ., 110 Cummington Mall, Boston, MA 02215, tmp@bu.edu)

Cavitation, or the creation and oscillation of bubbles, has a pivotal role in a variety of biomedical applications that involve ultrasound. For example, stable oscillating bubbles (stable cavitation) can reversibly permeabilize biological interfaces, including cell membranes and the blood-brain barrier. Bubbles that collapse inertially (inertial cavitation) radiate broadband emissions that can enhance ultrasound-mediated heating and thermal ablation. These cavitation-mediated bioeffects may provide medical benefit provided the location, type, and intensity of cavitation activity can be controlled. Microfluidic devices can be used to produce monodisperse microbubbles coated with viscoelastic shells that promote nonlinear oscillations. Liquid perfluorocarbon nanoemulsions can serve as long-circulating cavitation nuclei capable of extravasating into extravascular tissues. Using these technologies, cavitation nuclei can be tailored to provide a predictable acoustic response in a targeted location, thus providing a desired biological outcome.

Contributed Papers

2:00

3pBAb2. Correlation of cavitation activity with ultrasound-enhanced delivery of compounds to erythrocytes *ex vivo*. Emily M. Murphy, Mariah C. Priddy (BioEng., Univ. of Louisville, 2301 S Third St., Lutz Hall, Rm. 400, Louisville, KY 40292), Brett R. Janis, Michael A. Menze (Biology, Univ. of Louisville, Louisville, KY), and Jonathan A. Kopechek (BioEng., Univ. of Louisville, Louisville, KY, jakope01@louisville.edu)

Blood transfusions are one of the most common medical procedures in hospitals, but shortages of erythrocytes often occur due to their limited shelf life (6 weeks) when refrigerated. Preservation of erythrocytes in a dried state offers a potential solution to challenges faced with blood storage, and preservative compounds such as trehalose have been identified in organisms that survive desiccation in nature. However, these compounds do not readily cross mammalian cell membranes. Therefore, we are exploring the use of sonoporation to facilitate their delivery into erythrocytes *ex vivo*. In this study, we assessed the effect of cavitation activity on delivery of a fluorescent compound similar in size to trehalose (fluorescein) to human erythrocytes. Microbubbles were added to erythrocyte solutions and sonicated (2.5 MHz, 4 cycles) at various pressures and durations. Fluorescence was quantified with flow cytometry. The amplitude of broadband emissions in the first 8 seconds of sonication did not correlate with delivery ($r^2 = 0.23$), whereas after 8 seconds the broadband emissions amplitude was associated with increased delivery to erythrocytes ($r^2 = 0.97$). These results suggest that the timing of cavitation activity, rather than the amplitude alone, may be an important factor in ultrasound-mediated delivery of compounds into erythrocytes.

2:15

3pBAb3. Urinary stone erosion and fragmentation under low-intensity quasi-collimated ultrasound using gas-filled microbubbles with stone-targeting lipid shells. Yuri A. Pishchalnikov, William Behnke-Parks, Matt Mellema, Matt Hopcroft, Alice Luong (Applaud Medical, Inc., 953 Indiana St., San Francisco, CA 94107, yurapish@gmail.com), Tim Colonius, Kazuki Maeda (Dept. of Mech. and Civil Eng., California Inst. of Technol., Pasadena, CA), Kyle Morrison (Sonic Concepts, Inc., Bothell, WA), and Daniel Laser (Applaud Medical Inc., San Francisco, CA)

Urinary stone lithotripsy critically depends on the presence of cavitation nuclei at the stone surface. We hypothesized that introduction of stone-targeting microbubbles could increase cavitation activity at a stone surface sufficiently to allow stone erosion and fragmentation at peak negative pressures much lower than in acoustic energy-based urinary stone interventions with induced cavitation nuclei alone. Gas-filled microbubbles were produced with calcium-binding moieties incorporated into an encapsulating lipid shell. Stone surface coverage with these targeting microbubbles was found to approach an optimal (considering microbubble expansion during insonation) range of 5–15% with incubation times of three minutes or less. Using high-speed photomicroscopy, we observe bound microbubbles expanding 10- to 30-fold under insonation with quasi-collimated sources at mechanical indexes below 1.9. For observed stand-off parameters in the range of 0.2–0.6, the modeled collapse-generated shockwaves exceed 100 MPa. In swine model studies with these targeting microbubbles, stone fragmentation into passable fragments occurs with treatment times around 30 minutes, while post-treatment examination of ureters and kidneys shows no evidence of urothelium damage or renal parenchymal hemorrhage. The stone-targeting microbubbles reported on here have formed the basis for a new non-invasive urinary stone treatment which recently entered human clinical trials.

2:30

3pBAb4. Analytic prediction of histotripsy-induced bubble growth in an elastic medium. Kenneth B. Bader (Radiology, Univ. of Chicago, 5841 South Maryland Ave., MC 2026, Chicago, IL 60637, baderk@uchicago.edu)

Histotripsy is a form of therapeutic ultrasound that liquefies tissue mechanically via acoustic cavitation. Bubble growth due to histotripsy excitation has been calculated analytically with high accuracy in a fluid medium. Tissue elasticity is a determining factor in the therapeutic efficacy of histotripsy, and has not been considered in analytic bubble dynamics calculations. In this study, an analytic model to predict histotripsy-induced bubble expansion based on the medium surface tension, viscosity, and inertia was extended to include the effects of medium stiffness. Good agreement was observed between the predictions of the model and numerical computations. The predictions of the model were also consistent with experimental observation of bubble expansion, though are dependent on the elasticity form. Bubble growth was weakly dependent on the medium elasticity for highly nonlinear, shock scattering histotripsy pulses, but was strongly dependent for purely tensile, microtripsy pulses. For both forms of histotripsy, bubble growth was completely suppressed when the elastic modulus exceeded 20 MPa and the peak negative pressures was less than 50 MPa. These results highlight the importance of the histotripsy insonation scheme on bubble growth in an elastic medium, as well the range of tissue elasticities for efficacious bubble-induced liquefaction.

2:45

3pBAb5. On amplification of radial oscillations of microbubbles due to bubble-bubble interaction in polydisperse microbubble clusters under ultrasound excitation. Hossein Haghi, Amin Jafari Sojahrood, and Michael C. Kolios (Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2k3, Canada, hossein.haghi@ryerson.ca)

Optimizing the performance of microbubbles (MBs) in applications not only requires a good understanding of the dynamics of individual MBs, but also knowledge of their interactions with each other within polydisperse clusters. However, the behavior of acoustically driven MB clusters is not well understood. Most studies have been limited to cases of a few interacting MBs for a limited range of acoustical parameters. We use a numerical method to simulate the dynamics of clusters of (20–50 MBs) randomly distributed MBs. MBs with the size distribution of 2–8 micron were sonicated with pressures between 1 and 250 kPa, frequencies between 0.5 and 10 MHz and concentrations between 2e3 and 2.5e6 MBs/mL. Bifurcation structures of radial oscillations of each MB within a cluster were studied. Results show that in a polydisperse MB cluster, larger MBs (even in small numbers) can amplify the radial oscillations of smaller MBs. Amplification occurs if the resonance peaks of larger MBs coincide with the resonance peaks of smaller MBs. Additionally, there is an optimum number density at which amplification is maximized. Results suggest that the presence of bigger MBs may enhance the superharmonic oscillations of the smaller MBs which can be applied in superhamonic ultrasound imaging using MBs.

WEDNESDAY AFTERNOON, 9 MAY 2018 GREENWAY C/D, 2:15 P.M. TO 3:20 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

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

Chair's Introduction—2:15

Invited Papers

2:20

3pID3. An objective metric for describing the basic acoustics of binaural directivity patterns in humans. Andrew Dittberner, Changxue Ma (Res. Group, GN Adv. Sci., 2601 Patriot Blvd., Glenview, IL 60026, adittberner@gnresound.com), and Rob de Vries (Res. Group, GN Adv. Sci., Eindhoven, Netherlands)

It is accepted knowledge that having two ears are better than one when trying to listen to a signal of interest in the presence of spatially-separated noise sources (e.g. Blauert, 1997; Bregman, 1994; Zurek, 1993). Models have been proposed purporting of the benefits of the head shadow effect, binaural interactions, and cognitive factors that explain how one can understand sound with linguistic or other contextual meaning better in the presence of spatially-separate noise sources. However, less discussed is the attribute of human listeners having the ability to also hear and identify sound sources, seemingly on demand, that occur around them, a condition made possible by the fact of having two ears. Zurek (1993) proposed and discussed at length on the directivity effects of binaural listening (e.g., Better Ear Strategy). What is proposed in this study is an extension to this model to include the omni-directional directivity effects of binaural listening to describe the listener's ability to remain connected and aware of the sound landscape that surrounds them. Where the head shadow effect plays a role in improving the signal-to-noise ratio in one of the two ears, this strategy looks at how the two ears, due to their geometric location on the head, allows for the head to be acoustically transparent and keeps the listener connected to their surrounding sound landscape.

2:40

3pID2. Hot topics in structural acoustics and vibration: Advances in vibroacoustic modeling and novel materials. Robert M. Koch (Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil)

Two major areas of current research focus in Structural Acoustics and Vibration (SAV) are in (1) the development of advancements in the Physics-Based Modeling (PBM) and simulation of real-world, large-scale vibroacoustic systems and also in (2) the development and incorporation of novel engineered materials to improve structural acoustic response performance from a broad array of perspectives. This paper presents examples from a wide survey of recent SAV computational PBM research, from the inclusion of additional complex, coupled modeled phenomena, to improvements in modeling efficiency, accuracy, and/or full frequency spectrum response prediction, through computational advances in the ability to solve extremely large-scale, previously intractable, structural acoustics models. Related to the second materials-based SAV research concentration, recent advancements and innovations in SAV-relevant materials including acoustic metamaterials, zero- and negative-Poisson's ratio materials, single crystal ceramics, nanomaterials, etc., are also presented.

3:00

3pID1. Model-based Bayesian signal processing in acoustics. Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu)

Bayesian signal processing has been increasingly applied to a wide variety of acoustical research and engineering tasks. Bayesian probability theory provides acousticians with an elegant framework for inferential data analysis which facilitates learning from acoustic experimental investigations that provide an improved understanding of the underlying theory. In these inferential analysis tasks, certain prior knowledge is often available about the acoustical phenomena under investigation, based either on the underlying physical theory or on certain phenomenological relationships. Bayesian probability theory allows this available information to be incorporated in the processing and analysis and exploited in the Bayesian framework as physical or phenomenological models. Many analysis tasks in acoustics often include two levels of inference, the model selection and the parameter estimation. Bayesian signal processing provides solutions to these two levels of inference by extensively using Bayes' theorem within this unified framework. This talk will discuss various model-based approaches recently applied to signal processing and analysis in acoustics using either one or both levels of inference.

WEDNESDAY AFTERNOON, 9 MAY 2018 GREENWAY J, 1:00 P.M. TO 3:05 P.M.

Session 3pPA

Physical Acoustics, Biomedical Acoustics and Psychological and Physiological Acoustics: Ultrasound and High Frequency Sound in Air in Public and Work Places: Applications, Devices, and Effects

Timothy Leighton, Cochair
Craig N. Dolder, Cochair
Institute of Sound and Vibration Research, University of Southampton, Highfield Campus, Southampton SO17 1BJ, United Kingdom

Chair's Introduction—1:00

Invited Papers

1:05

3pPA1. Ultrasound, human health, safe levels, and Cuba: What do we know? Timothy Leighton (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton, Hampshire SO17 1BJ, United Kingdom, tgl@soton.ac.uk)

While definitive statements are impossible, a slim evidence base suggests the following on weight-of-probabilities regarding ultrasonic adverse effects in humans. One study at extremely high intensities reports physical effects (notably burning between fingers/nostrils). At lower intensities, adverse psychological effects occur in only a subset of the population (susceptibility possibly decreasing with age), are restricted to frequencies below ~25 kHz, probably result from the extraordinary sensitivity of our hearing/balance systems, and can be difficult to separate from (and may be causally related to) adverse effects of anxiety and annoyance. This does not remove the need for protection, especially for increasingly common public exposures, where the exposure and exposed person are often uncharacterised. Only one interim guideline from 1984 addresses maximum permissible levels (MPLs) for public exposure. It is based on scant

evidence, and may or may not be appropriate. All other guidelines relate to occupational exposure. These MPLs are a legacy of decades of copying previous guidelines, which were themselves based on inadequate sampling (usually a small cohort of adult men), and averaging practices which obscured the particular sensitivities of a subset of the population. Against this background, the likelihood, or not, of an ultrasonic weapon in Cuba will be discussed.

1:25

3pPA2. Adverse effects of very high-frequency sound and ultrasound on humans. Mark D. Fletcher (Inst. of Sound and Vib. Res., Univ. of Southampton, 15 Consort Rd., Eastleigh So50 4JD, United Kingdom, mdf1f15@soton.ac.uk), Sian Lloyd Jones (Dept. of Audiol. and Hearing Therapy, Royal South Hants Hospital, Southampton, United Kingdom), Craig N. Dolder, Paul White, Timothy Leighton, and Ben Lineton (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, Hampshire, United Kingdom)

For many years, workers have reported adverse symptoms resulting from exposure to very high-frequency sound (VHFS) and ultrasound (US), including annoyance, dizziness and difficulty concentrating. Recent work showing the presence of a new generation of VHFS/US sources in public places has reopened the debate about whether adverse effects can be caused by exposure to VHFS/US. Our field measurements of VHF/US sources in public places have identified devices producing up to 100 dB SPL at 20 kHz. Nearly all of the sources measured, including those in places occupied by tens of millions of people each year, are likely to be clearly audible to many young people. We have conducted two studies. The first looked at adverse symptoms resulting from exposure to audible VHFS/US, and the second was a double-blind study of adverse symptoms resulting from exposure to inaudible VHFS/US. In each study, both symptomatic participants, who reported previously experiencing symptoms, and asymptomatics participants, who did not, were tested. We found evidence that symptoms were produced by exposure to audible VHFS/US but not by inaudible sound. It is possible that the substantial effects reported for inaudible VHFS/US exposure were not reproduced because of ethical restrictions on stimulus level and duration.

1:45

3pPA3. Measurement of ultrasound radiated from rodent repellents used in an occupational space, and auditory evaluation of the sound. Mari Ueda (Information Technol., Gent Univ., 4-9-1 Shiobaru, Fukuoka, Minami-ku 815-8540, Japan, m-ueda@design.kyushu-u.ac.jp)

In order to clarify the emission levels and hearing status of ultrasound used in an occupational spaces by pest control companies, three surveys were performed. The results of the acoustic field measurement showed that the rodent repellent device had a peak level and frequency of approximately 100 dB and 19 kHz, respectively. The results of an auditory evaluation experiment with 51 adult workers showed that younger workers recognized the ultrasound from the electronic rodent repellent device more clearly than the elderly workers.

2:05

3pPA4. Ultrasound measurements in the work environment. Jan Radosz and Dariusz Pleban (Central Inst. for Labour Protecion - National Res. Inst., Czerniakowska, 16, Warsaw 00-701, Poland, jarad@ciop.pl)

For the frequency range above 20 kHz, there is no clear and complete information on the factors influencing the result of a measurement of sound pressure level. Moreover, there are no current international standards for performing measurements of ultrasound at work stations. The authors presents a possibility for the adaptation of the existing measurement methods, in particular, the requirements for measuring instruments, procedures to be followed while performing measurements, the application of a correction to measurement results, and the determination of measurement uncertainty. The development of a consistent method of ultrasound measurement is of utmost importance in carrying out an assessment and reducing the risk of exposure to this physical factor in the work environment.

2:25

3pPA5. Low power wireless communication between personal electronic devices and hearing aids using high frequency audio and ultrasound. Jonathon Miegel (ARC Training Ctr. in Biodevices, Swinburne Univ. of Technol., 545 Burwood Rd, Hawthorn, VIC 3122, Australia, jmiegel@swin.edu.au), Philip Branch (Dept. of Telecommunications, Elec., Robotics and Biomedical Eng., Swinburne Univ. of Technol., Hawthorn, VIC, Australia), and Peter Blamey (Blamey Saunders hears, East Melbourne, VIC, Australia)

Hearing aids continue to be the main intervention for hearing loss but their ease of use and control is of concern due to their small size. While technological advances in Bluetooth Low Energy have allowed for improved wireless control, in particular, between personal electronic devices, its use for communication with hearing aids is problematic due to their limited battery life. This research investigates the implementation of acoustic wireless communication between personal electronic devices and hearing aids using multiple modulation schemes utilizing frequencies between 16 and 20 kHz. The performance of each modulation scheme is assessed over a 3 metre range and the power consumption compared to that of Bluetooth Low Energy.

2:45

3pPA6. A scavenger hunt using ultrasonic geocaches. Craig N. Dolder (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield Ave., Southampton, Hampshire SO17 1BJ, United Kingdom, dolder@utexas.edu)

Sound is commonly used for either communication or navigation. An ultrasonic scavenger hunt was designed that does both and is designed to raise awareness about acoustics. This scavenger hunts utilizes ultrasonic geocaches to both give information to the participants and educate them on topics including, the fact that sound may not be audible, the concept of hearing loss, other animals hear at different frequencies, and general facts from the hosting event. The geocaches use the frequency band above typical human hearing but still within the bandwidth of most personal electronics, 20 kHz–22 kHz. This band can be picked up by common smartphones and tablets and viewed using free spectrogram applications. The maximum sound pressure level output by the geocache devices falls below maximum public exposure recommendations but the signal is still visible on a spectrogram. The scavenger hunt was trialed at a science engagement event at the University of Southampton with over 6000 in attendance.

Session 3pPP

Psychological and Physiological Acoustics and Speech Communication: Acoustics Outreach to Budding Scientists: Planting Seeds for Future Clinical and Physiological Collaborations '18

Elin Roverud, Cochair
Dept. of Speech, Language & Hearing Sciences, Boston University, 635 Commonwealth Avenue, Boston, MA 02215

Anna C. Diedesch, Cochair
Communication Sciences & Disorders, Western Washington University, 516 High St., MS 9171, Bellingham, WA 98225

Chair's Introduction—1:10

Invited Papers

1:15

3pPP1. Auditory, cognitive, and linguistic processing skills in individuals with hearing loss. Shivali Appaiah Konganda (Dept. of Linguist, Macquarie Univ., Ground Fl., Australian Hearing Hub, 16 University Ave., Sydney, NSW 2109, Australia, shivali.appaiah-konganda@students.mq.edu.au), Mridula Sharma, Jessica J. Monaghan (Dept. of Linguist, Macquarie Univ., North Ryde, NSW, Australia), Gitte Keidser, Joaquin Tomas Valderrama Valenzuela (National Acoust. Labs., Australian Hearing, Australia, North Ryde, NSW, Australia), John Newall (Dept. of Linguist, Macquarie Univ., North Ryde, NSW, Australia), and Elizabeth Beach (National Acoust. Labs., Australian Hearing, Australia, North Ryde, NSW, Australia)

Hearing impairment affects a person's ability to communicate effectively. People with hearing loss (HL) report difficulty communicating in noise, even when the HL is compensated by conventional amplification. This study aims to investigate factors that contribute to understanding speech in noise. Nine adults with HL and nine controls participated in the study. The test-battery include auditory, cognitive and linguistic tests. For the HL group, auditory stimuli were filtered with NAL-RP prescription to compensate for their HL. Results indicate a significant difference in performance between the groups on the Modulation Detection Threshold (MDT) test [F (1, 15) = 3.24, p =0.04] and the speech recognition in noise test [f (1, 15) = 25.6, p<0.001]. HL group performed better on the MDT and poorer at recognising speech in noise possibly due to broadening of auditory filters. With the broadened auditory filters in mind, this result supports the fact that they would have poor frequency specificity, detrimental for speech recognition. HL group performed better than the control group on the cognitive spare capacity test [f (1, 15) = 4.72, p= 0.04]. Preliminary data suggests that adults with HL may compensate for hearing-related difficulties in certain situations by using their cognitive skills.

1:35

3pPP2. Musical emotion recognition in bimodal patients. Kristen D'Onofrio (Vanderbilt Univ., 1215 21st Ave. S., Nashville, TN 37232, kristen.l.donofrio@vanderbilt.edu), Charles Limb, Meredith Caldwell (UCSF, San Francisco, CA), and René Gifford (Vanderbilt Univ., Nashville, TN)

Several cues are used to convey musical emotion, the two primary being musical mode and musical tempo. Specifically, major and minor modes are regarded as having positive and negative valence, respectively, and songs at fast tempi (i.e., quarter note = 92-196, Gosselin *et al.* (2005); quarter note = 80-255, Peretz *et al.* (1998)) are associated with more positive valence compared to songs at slow tempi (i.e., quarter note = 40-60, Gosselin *et al.* (2005); quarter note = 20-100, Peretz *et al.* (1998)). Caldwell *et al.* (2015) demonstrated that CI users relied on tempo (fast vs. slow) with little regard for mode (major vs. minor) when interpreting musical emotion, a finding consistent with spectral cues being poorly represented in CI users. The current study is an extension of Caldwell *et al.* (2015) examining how mode and tempo cues impact musical emotion recognition in bimodal listeners. Our primary hypothesis is that, unlike CI only users, bimodal listeners—with access to F0 and fine structure—will utilize both mode and tempo cues in a manner similar to normal-hearing listeners. Our secondary hypothesis is that low-frequency spectral resolution in the contralateral ear (via tuning curves and spectral modulation detection) will be significantly correlated with degree of bimodal benefit for musical emotion perception.

1:55

3pPP3. Perceptual and neural representation of consonants in hearing impaired listeners. Yaqing Su (Starkey Hearing Res. Ctr., 44 Cummington St, Boston, MA 02215, ysu11@bu.edu), Narayan Sankaran (Starkey Hearing Res. Ctr., Sydney, New South Wales, Australia), and Jayaganesh Swaminathan (Starkey Hearing Res. Ctr., Berkeley, CA)

Difficulties in perception of consonants have been associated with speech perception difficulties in hearing-impaired (HI) listeners. However, the neural bases underlying such difficulties have not been clearly explored and understood. The goal of this study was to use scalp electroencephalography (EEG) recordings to better understand the neural mechanisms that contribute to the consonant perception

difficulties in HI listeners. Such a psychophysiological approach can lead to the development of improved hearing-aid fitting and speech enhancement strategies. Perceptual and EEG responses to vowel-consonant-vowel speech were measured in 8 HI listeners with and without amplification. A machine-learning classifier was trained to discriminate the EEG signal evoked by each consonant. The performance of the classifier was compared to the HI listeners' psychophysical performance. For all subjects, consonant intelligibility was better in aided compared to unaided listening condition, but overall performance was well below ceiling. An information transmission analysis showed that place and manner of articulation were more affected than voicing and nasality. EEG waveform showed different response patterns for each consonant, and the machine-learning classifier was able to successfully "decode" consonants from the EEG signal. However, a straightforward relationship between the neural and perceptual representation of consonants could not be established in HI listeners.

2:15

3pPP4. Characterizing cortical auditory networks in bilateral cochlear implant users using electroencephalography. Daniel Smieja (Univ. of Toronto, The Hospital for Sick Children, 555 University Ave., Toronto, ON M5G1X8, Canada, daniel.smieja@mail.utoronto.ca), Benjamin Dunkley, Blake Papsin, and Karen Gordon (The Hospital for Sick Children, Toronto, ON, Canada)

Objective: This research aims to characterize the cortical network involved in listening with bilateral cochlear implants (CIs) through measures of functional connectivity. *Rationale*: A better understanding of the underlying cortical network involved in hearing will help elucidate remaining challenges experienced by children who received bilateral cochlear implants early relative to their peers with normal hearing. This work extends our previous focus on plasticity in temporal regions to include multiple cortical areas and considers the phase relationships between neural source signals. *Methods*: The data were collected using 64-channel electroencephalography in response to click stimuli in a passive listening condition. Source reconstruction was applied using the TRACS beamformer in order to localize the underlying neural generators. Sources were mapped to regions in the Automated Anatomical Labeling atlas and connectivity analyses were performed to examine statistical relationships between the atlas regions in order to generate network models. *Results*: Preliminary data show that in addition to the primary auditory areas, regions in the precuneus and frontal areas are activated in response to sound. The connectivity analyses are ongoing. *Significance*: Differences in the cortical auditory networks will help to understand how the auditory system is integrated in the cortex in bilateral cochlear implant users.

2:35

3pPP5. Tone-evoked acoustic change complex (ACC) in an animal model. Alessandro Presacco and John C. Middlebrooks (Otolaryngol., Univ. of California, Irvine, Medical Sci. D, Rm. D404, Irvine, CA 92697, presacca@uci.edu)

The auditory change complex (ACC) is a cortical evoked potential complex generated in response to a change (e.g., frequency or level) within an ongoing auditory stimulus. The ACC has been recorded in both normal-hearing human subjects and in cochlear implant users, suggesting that the ACC would be useful in clinical applications. Here, we investigate the feasibility of recording ACC in response to frequency or level changes in sedated cats. Five purpose-bred cats were sedated with ketamine and acepromazine. Continuous tones alternated between high and low frequencies or levels in 500-ms blocks. Frequency and level steps were varied parametrically. Scalp potentials were recorded with needle electrodes (two active electrodes = one on each hemisphere, reference: mastoid, ground = back of the cat). ACC was successfully elicited in all cats by both frequency and level steps. In many cases, ACCs were markedly greater for increasing or for decreasing stimulus steps. The discrimination thresholds measured were in good agreement with previous behavioral studies in which cats where trained to perform similar frequency or level discriminations. The results indicate that the ACC will be a useful tool for evaluating novel acoustical or electrical stimulation modes that are not yet feasible in humans.

2:55

3pPP6. Activity in human auditory cortex represents spatial separation between concurrent sounds. Martha Shiell, Lars Hausfeld, and Elia Formisano (Dept. of Cognit. Neurosci., Maastricht Univ., Oxfordlaan 55, Maastricht 6229 EV, Netherlands, marthashiell@gmail.com)

Auditory spatial information can enhance stream segregation in an auditory scene, but little is known about how auditory space is represented in the human cortex. If spatial cues are used for scene analysis, then it is the distance between sounds rather than their absolute positions that is essential, and thus, we hypothesized that auditory cortical neurons encode separation between sounds rather than absolute location. To test this hypothesis, we measured human brain activity in response to spatially-separated concurrent sounds with magnetic resonance imaging at 7 Tesla. Stimuli were spatialized amplitude-modulated broadband noises, recorded for each participant via in-ear microphones prior to scanning. Using a linear support vector machine classifier, we investigated if sound location and/or spatial separation between sounds could be decoded from the activity in Heschl's gyrus and the planum temporale. The classifier was successful only when comparing patterns associated with the conditions that had the largest difference in perceptual spatial separation, and not in conditions where only location changed and separation remained constant. Our pattern of results suggest that the representation of separation is not merely the combination of single locations, but is an independent feature of the auditory scene.

Session 3pSA

Structural Acoustics and Vibration, Signal Processing in Acoustics and Noise: Model Reduction for Structural Acoustics and Vibration

Kuangcheng Wu, Cochair
Naval Surface Warfare Center - Carderock, 9500 MacArthur Blvd, West Bethesda, MD 20817

Hubert S. Hall, Cochair
Mechanical Engineering, The Catholic University of America, 620 Michigan Ave. NE, Washington, DC 20064

Chair's Introduction—1:30

Invited Papers

1:35

3pSA1. Applications of the hybrid finite element method. Nickolas Vlahopoulos and Sungmin Lee (Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48109, nickvl@umich.edu)

The hybrid finite element analysis (Hybrid FEA) method is based on combining conventional finite element analysis (FEA) with energy finite element analysis (EFEA) for expediting the FEA computations when very dense models are needed. The difficulty in using conventional FEA at higher frequencies originates from requiring a very large number of elements in order to capture the flexible wavelength of the panel members which are present in a structure. In the Hybrid FEA the conventional FEA model is modified by de-activating the bending behavior of the flexible panels in the FEA computations and introducing instead a large number of dynamic impedance elements for representing the omitted bending behavior. The excitation is considered to be applied on the conventional FEA model and the vibration analysis is conducted. The power flow through the dynamic impedance elements is computed and applied as excitation to the EFEA model of the flexible panels. The EFEA analysis computes the vibration of the flexible panels. In the past, the Hybrid FEA has been utilized successfully for evaluating the vibration is production automotive and rotorcraft structures. A brief theoretical background will be reviewed, the practical aspects of the method will be discussed, and results from previous correlation studies will be presented.

1:55

3pSA2. A symmetrical formulation of coupled BEM/FEM for submerged structures based on acoustical reciprocity. Pei-Tai Chen (Dept. of System Eng. and Naval Architecture, National Taiwan Ocean Univ., No.2,Pei-Ning Rd. Keelung 20224, Keelung Keelung 20224, Taiwan, ptchen@mail.ntou.edu.tw)

The paper presents a structural acoustics formulation for elastic structures submerged in water. The structural equation is described by a finite element method where the linear displacement variables on the wetted surface are chosen locally as one normal displacement and the other two displacements tangent to the wetted surface, whereas the rest of rotational displacements or degrees of freedom not contacting with water are defined globally. A boundary element formulation describing the acoustic loading on the structure is expressed as a function of normal velocity or displacement on the wetted surface. The coupling of the FEM and BEM is through the normal velocity, or equivalently, the normal displacement on the wetted surface. An acoustical reciprocity is used to prove that the associated acoustic loading expressed as the normal displacement of the wetted structure is a complex symmetric matrix. This matrix can be viewed an acoustic element whose degrees of freedom are the normal displacements of the wetted surface. Thus, the coupled FEM and BEM becomes a symmetric banded matrix formulation, leading to an efficient numerical way to solve the equation. A capped cylindrical shell with periodical ring stiffeners and bulkheads submerged in water is used to demonstrate the present numerical method.

2:15

3pSA3. Finite element computation of the scattering response of objects buried in ocean sediments. Anthony L. Bonomo (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd, West Bethesda, MD 20817, anthony.bonomo@gmail.com) and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

To assist in locating objects at or near the ocean floor, there is a need for computational models that can predict the scattering response of objects fully or partially buried in ocean sediments such as sand. In reality, sand can best be understood as a two-phase porous medium and the shear waves supported by the sediment can contribute meaningfully to the acoustic signature of the buried objects. Due to the slow phase speeds of these shear waves, the computational burden of finite element models can be excessive. In this talk, efficient modeling techniques and methods for reducing the computational time of these models are covered. [Work supported by ONR, Ocean Acoustics.]

2:35

3pSA4. A procedure for ship identification through acoustic signal processing. Gee-Pinn J. Too (Dept. of Systems and Naval Mechatronic Eng., National Cheng Kung Univ., No. 1 University Rd. NCKU, Tainan 70101, Taiwan, toojames@yahoo.com)

All ship's mechanical structures are different. Hence, the radiated noise spectra are different, especially in line spectra. Line spectra are composed of the spectra due to shaft speed and structure resonance. It is appropriate to identify ship by line spectra. In this study, a ship noise model has been used by combining line spectrum and continuous spectrum of a ship. Then, artificial neural network (ANN) has been used to calculate the contents of the line spectrum of a ship noise in order to identify a ship. Furthermore, in order to identify ship correctly, the received signal is restored to the source signal. Before signal restoration, the source needs to be located. Hence the signal restoration and source localization are the important issues for ship identifications. Finally, adaptive time reversal (ATR) is used to locate source and to restore signal. The results indicate that ATR is effective in source localization and source signal restoration. Therefore, it shows great advantages in identifying a ship

2:55

3pSA5. Extraction of independent non-Gaussian components from non-Gaussian background. Zbyněk Koldovský (Inst. of Information Technol. and Electronics, Tech. Univ. of Liberec, Studentska 2, Liberec 46117, Czechia, zbynek.koldovsky@tul.cz)

Independent component analysis (ICA) is a popular method for Blind Signal Separation applied to multichannel linear recordings. Convolutive mixtures of acoustic signals can be separated by applying ICA in the frequency domain to each sub-band separately or jointly as in Independent Vector Analysis (IVA). However, this way, the mixtures are separated into that many components as is the number of sensors. ICA and IVA are therefore not effective in applications where only one (or few) signals are of practical interest. We introduce a new parameterization of the ICA mixing model that is optimized for the extraction of one signal of interest (SOI). The approach is called Independent Component Extraction (ICE) and is closely related to methods for Blind Signal Extraction (BSE) such as One-unit FastICA. However, BSE methods assume that background signals (the other signals than SOI) are all Gaussian, which limits their accuracy. In this work, we show that ICE can be extended also for a non-Gaussian background so that the accuracy of algorithms is improved, although all components are not separated as in ICA. We also introduce an extension for the extraction of a vector component, so-called Independent Vector Extraction.

WEDNESDAY AFTERNOON, 9 MAY 2018 — NICOLLET A, 1:00 P.M. TO 3:20 P.M.

Session 3pSC

Speech Communication: Tools and Technology for Speech Research (Poster Session)

Mary M. Flaherty, Chair
Psychology, SUNY Buffalo, 392 Park Hall, North Campus, Buffalo, NY 14260

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 3:20 p.m.

Contributed Papers

3pSC1. Fearless steps: Advancing speech and language processing for naturalistic audio streams from Earth to the Moon with Apollo. John H. L. Hansen, Abhijeet Sangwan, Lakshmish Kaushik, and Chengzhu Yu (Jonsson School of Eng. & Comput. Sci., CRSS: Ctr. for Robust Speech Systems; UTDallas, CRSS: Ctr. for Robust Speech Systems, The University of Texas at Dallas; 800 W Campbell Rd., Richardson, TX 75080-3021, john.hansen@utdallas.edu)

NASA's Apollo program represents one of the greatest achievements of mankind in the 20th century. CRSS-UTDallas has completed an effort to digitize and establish an Apollo audio corpus. The entire Apollo mission speech data consists of well over ~100,000 hours. The focus of this effort is to contribute to the development of Spoken Language Technology based algorithms to analyze and understand various aspects of conversational speech. Towards achieving this goal, a new 30 track analog audio decoder was designed using NASA Soundscriber. We have digitized 19,000 hours of data from Apollo 11,13,1 missions: named "Fearless Steps". An automated diarization and transcript generation solution was developed based on deep neural networks (DNN) automatic speech recognition (ASR) along with Apollo mission specific language models. Demonstration of speech technologies including speech activity detection (SAD), speaker identification (SID), and ASR are shown for segments of the corpus. We will release this corpus to the SLT community. The data provide an opportunity for challenging tasks in various SLT areas. We have also defined and proposed 5 tasks as a part of a community based SLT challenge. The five challenges are as follows: (1) automatic speech recognition, (2) speaker identification, (3) speech activity detection, (4) speaker diarization, and (5) keyword spotting and joint topic/sentiment detection. All data, transcripts, and guidelines for employing the fearless steps corpus will be made freely available to the community.

1:00

3pSC2. UTDallas-PLTL:Advancing multi-stream speech processing for interaction assessment in peer-led team learning. John H. L. Hansen, Harishchandra Dubey, Abhijeet Sangwan, Lakshmish Kaushik, and Vinay Kothapally (Jonsson School of Eng. & Comput. Sci., CRSS: Ctr. for Robust Speech Systems, UTDallas, The University of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, john.hansen@utdallas.edu)

Robust speech processing for single-stream audio data has achieved significant progress in the last decade. However, multi-stream speech processing poses new challenges not present in single-stream data. The peer-led team learning (PLTL) is a teaching paradigm popular among US universities for undergraduate education in STEM courses. In collaboration with UTDallas Student Success Center, we collected CRSS-PLTL and CRSS-PLTL-II corpora for assessment of speech communications in PLTL sessions. Both corpora consist of longitudinal recordings of five teams studying undergraduate Chemistry and Calculus courses consisting of 300 hours of speech data. The multi-stream audio data has unique challenges: (i) time-synchronization; (ii) multi-stream speech processing for speech activity detection, speaker diarization and linking, speech recognition, and (iii) behavioral informatics. We used a 1 kHz tone at the start and end of each session for time-synchronization of multi-stream audio. We leveraged auto-encoder neural network for combining MFCC features from multiple streams into compact bottleneck features. After diarization, each speaker segment is analyzed for behavioral metrics such as (i) dominance; (ii) curiosity in terms of question inflections; (iii) speech rate; (iv) cohesion; and (v) turn-duration and turn-taking patterns. Results are presented on individual and team based conversational interactions. This research suggests new emerging opportunities for wearable speech systems in education research.

3pSC3. Estimation of emotional arousal from speech with phase-based features. Igor Guoth, Sakhia Darjaa, Marian Trnka, Milan Rusko, Marian Ritomský (Inst. of Informatics, Slovak Acad. of Sci., Dubravska cesta 9, Bratislava 845 07, Slovakia, igor.guoth@savba.sk), and Roman Jarina (Univ. of Zilina, Zilina, Slovakia)

The most commonly adopted approaches in speech emotion recognition (SER) utilize magnitude spectrum and nonlinear Teager energy operator (TEO) based features while information about phase spectrum is often omitted. The information about phase has been frequently overlooked in approaches applied by speech processing researchers due to the signal processing difficulties. We present study of two phase-based features: The relative phase shift (RPS) based features and modified group delay features (MODGDF) that represents phase structure of speech in the task of emotional arousal recognition. The evaluation is performed on the CRISIS acted speech database which allows us to recognize five levels of emotional arousal from speech. To exploit these features, we employ concept of deep neural network. The efficiency of the approaches based on features mentioned earlier is compared to baseline platform using Mel frequency cepstral coefficients (MFCCs) and all pole group delay features (APGD). The combination of another phase-based types of features with our baseline platform led to the overall improvement of performance of the system for different levels of emotional arousal. These results confirm that combination of phase information and magnitude information leads to the overall improvement of performance of such system and also that combination of different types of features representing phase information brings additional increment of the performance.

3pSC4. The speakers in the room corpus. Aaron Lawson (SRI Int., 333 Ravenswood Ave., Menlo Park, CA 94025, aaron.lawson@sri.com), Karl Ni (Lab41, Menlo Park, CA), Colleen Richey, Zeb Armstrong, Martin Graciarena (SRI Int., Menlo Park, CA), Todd Stavish, Cory Stephenson, Jeff Hetherly, Paul Gamble, and Maria Barrios (Lab41, Menlo Park, CA)

The speakers in the room (SITR) corpus is a collaboration between Lab41 and SRI International, designed to be a freely available data set for speech and acoustics research in noisy room conditions. The main focus of the corpus is on distant microphone collection in a series of four rooms of different sizes and configurations. There are both foreground speech and background adversarial sounds, played through high-quality speakers in each room to create multiple, realistic acoustic environments. The foreground speech is played from a randomly rotating speaker to emulate head motion. Foreground speech consists of files from LibriVox audio collections and the background distractor sounds will consist of babble, music, HVAC, TV/radio, dogs, vehicles, and weather sounds drawn from the MUSAN collection. Each room has multiple sessions to exhaustively cover the background foreground combinations, and the audio is collected with twelve different microphones (omnidirectional lavalier, studio cardioid, and piezoelectric) placed strategically around the room. The resulting data set was designed to enable acoustic research on event detection, background detection, source separation, speech enhancement, source distance, sound localization, as well as speech research on speaker recognition, speech activity detection, speech recognition, and language recognition.

3pSC5. Creation and characterization of an emotional speech database. Peter M. Moriarty, Michelle Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, moriarty@psu.edu), Rachel Wolf (Psych., Penn State Univ., University Park, PA), Rick Gilmore, and Pamela Cole (Psych., Penn State Univ., State College, PA)

Paralinguistic features of speech communicate emotion in the human voice. In addition to semantic content, speakers imbue their messages with prosodic features comprised of acoustic variations that listeners decode to extract meaning. Psychological science refers to these acoustic variations as *affective prosody*. Most studies of affective prosody obscure semantic content, although the resulting stimuli are less representative of naturally occurring emotional speech. The presented works details the creation of a naturalistic emotional speech database on which both acoustical analysis and a listening study were conducted. To this end, 55 adults were recorded speaking the same semantic content in happy, angry, and sad voices. Based on prior acoustic analyses of affective prosody, classic parameters were extracted including pitch, loudness, timing, as well as other low-level descriptors, and compared the acoustic features of each emotion with published evidence and theory. Preliminary results indicate that this naturalistic speech samples yielded acoustic features that are congruent with prior experimental stimuli of anger and happiness, but was less consistent with sadness. The results of the listening study indicated that listeners discriminated the intended emotions with 92% accuracy. The dataset therefore yielded a database of emotionally salient acoustical information for further analyses. [Work supported by NIH-R21-104547.]

3p WED. PM

3pSC6. Toward the automatic detection of manually labeled irregular pitch periods. Olivia Murton (Speech and Hearing BioSci. and Technol., Harvard Med. School, 1 Bowdoin St, #11, Boston, MA 02114, omurton@g.harvard.edu), Jeung-Yoon Choi (Massachusetts Inst. of Technol., Cambridge, MA), Daryush Mehta (Speech and Hearing BioSci. and Technol., Harvard Med. School, Boston, MA), and Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA)

Irregular pitch periods (IPPs) occur in a wide variety of speech contexts and can support automatic speech recognition systems by signaling word boundaries, phrase endings, and certain prosodic contours. IPPs can also provide information about emotional content, dialect, and speaker identity. The ability to automatically detect IPPs is particularly useful because accurately identifying IPPs by hand is time-consuming and requires expertise. In this project, we use an algorithm developed for creaky voice analysis by Kane *et al.* (2013) incorporating features from Ishi *et al.* (2008) to automatically identify IPPs in recordings of speech from the American English Map Task database. Short-term power, intra-frame periodicity, inter-pulse similarity, subharmonic energy, and glottal pulse peakiness measures are input into an artificial neural network to generate frame-by-frame creak probabilities. To determine a perceptually relevant threshold probability, the creak probabilities are compared to IPPs hand-labeled by experienced raters. Preliminary results yielded an area under the receiver operating characteristic curve of 0.82. Thresholds above 0.1 produced very high specificity, but even lower thresholds yielded fairly high sensitivity and specificity. These results indicate generally good agreement between hand-labeled IPPs and automatic detection, calling for future work investigating effects of linguistic and prosodic context.

3pSC7. Duration of connected speech needed to accurately estimate the articulatory-acoustic vowel space of a reading passage. Jason A. Whitfield, Anna Gravelin, Zoe Kriegel (Commun. Sci. and Disord., Bowling Green State Univ., 200 Health and Human Services Bldg., Bowling Green State University, Bowling Green, OH 43403, jawhitf@bgsu.edu), and Daryush Mehta (Dept. of Surgery–MGH, Harvard Med. School, Boston, MA)

Unlike other vowel space metrics, the articulatory-acoustic vowel space (AAVS) is calculated from the generalized variance of continuously sampled formant (F1-F2) traces. Given a sufficiently long speech sample, the AAVS should stabilize, though the duration required for the measure to converge remains unknown. The current investigation aimed to determine the amount of formant data needed for the AAVS to stabilize. Formant traces (20 ms frames, 10 ms intervals) were extracted using a Kalman-based autoregressive approach from readings of the Caterpillar passage produced by 16 speakers using habitual speech. The absolute percent difference (error) between cumulative AAVS estimates and the passage AAVS was calculated by iteratively adding F1-F2 pairs from successive voiced frames. Power functions were fit to the error plots using a least absolute residuals method to determine the frame at which the upper bound of the function fell within 5% of the passage AAVS. Across all speakers, the AAVS converged within 1065 voiced frames (i.e., 10.65 seconds of voiced speech). In terms of absolute speaking duration, all samples converged within 18 seconds (mean = 8.26; SD = 5.04). These data suggest that 15 to 20 seconds of connected speech is sufficient to provide a reasonable estimate of working vowel space using the AAVS.

3pSC8. Simulation of computerized screening for phonological errors in spoken utterances. Amitava Biswas, Anita Thames, and Steven Cloud (Speech and Hearing Sci., Univ. of Southern MS, 118 College Dr. #5092, USM-CHS-SHS, Hattiesburg, MS 39406-0001, Amitava.Biswas@usm.edu)

Fast and reliable diagnosis of phonological errors is often necessary. This presentation describes a simulation of computerized screening for phonological errors in spoken utterances. A set of common words will be included in a database. Every word in the database will be linked to several other words according to minimal pairs. For example, CAT will be linked to KATE, CUT, CAB, CASH, KITE, BAT, FAT, HAT, MAT, RAT, etc, whereas, RAT will be linked to CAT, RATE, WRITE, RASH, RAM, RAN, RACK, RAG, RAP, etc., and RAN will be linked to RAT, PAN, RAM, BAN, CAN, RUN, RAIN, etc. A set of hypothetical patterns of phonological errors in articulation will also be included in the database. The simulation will explore efficacy of navigating through the data base with the help of a custom software in predicting and correctly identifying the particular patterns of phonological errors.

3pSC9. Google speech recognition of an English paragraph produced by Korean college students in clear or casual speech styles. Byunggon Yang (English Education, Pusan National Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

These days voice models of automatic speech recognition software are sophisticated enough to process the natural speech of people without any previous training. However, not much research has reported on the use of speech recognition tools in the field of pronunciation education. This paper examined Google speech recognition of a short English paragraph produced by Korean college students in clear and casual speech styles in order to diagnose and resolve students' pronunciation problems. Thirty three Korean college students participated in the recording of the English paragraph. The Google soundwriter was employed to collect data on the word recognition rates of the paragraph. Results showed that the total word recognition rate was 73% with a standard deviation of 9.1%. The word recognition rate of clear speech was around 77.3% while that of the casual speech amounted to 68.7%. The reasons for the low recognition rate of casual speech were attributed to both individual pronunciation errors and the software itself as shown in its fricative recognition. Various distributions of unrecognized words were observed depending on each participant and proficiency groups. From the results, the author concludes that the speech recognition software is useful to diagnose each individual or group's pronunciation problems.

3pSC10. Automated formant tracking using reassigned spectrograms. Sean A. Fulop (Linguist, California State Univ. Fresno, 5245 N Backer Ave., PB92, Fresno, CA 93740-8001, sfulop@csufresno.edu) and Christine H. Shadle (Haskins Labs., New Haven, CT)

The measurement and tracking of formant frequencies is usually accomplished with a linear prediction spectrum estimate, combined with a heuristic-based tracking algorithm. Despite the broad acceptance of this procedure over decades, there is a known bias towards the nearest harmonic, and formant estimates are less accurate for speech with higher fundamental frequency. Previous research has established the superior accuracy of formant measurement using the reassigned spectrogram; however, the measurements must be done by hand, severely limiting its usefulness. Here we present a technique that can locate formants automatically in a reassigned spectrogram. First, the reassigned spectrogram of the signal must be heavily "pruned" so that only a few points remain which highlight the detected frequency components, which are the presumed formants. Second, we apply a simple ridge-finding routine to the pruned spectrogram, to determine the formant tracks automatically. This process was tested on speech samples—/hVd/-words—by four speakers previously used in a comparison of 5 automatic algorithms and manually-measured reassigned spectrograms, and on breathy and falling tones in a speaker of Hmong. The results of this process are generally as good or better than the manual measurement when the optimal parameters for ridge-finding are used.

3pSC11. Effects of a mask rim leak on AC airflow measurement. Nicholas A. May and Ronald Scherer (Commun. Sci. & Disord., Bowling Green State Univ., 103 South Main St., Apt. 10, Bowling Green, OH 43402, nmay@bgsu.edu)

Studies in airflow during speech production typically use a pneumotachographic mask system to measure expired airflows. Accurate measures of airflow using the Glottal Enterprises MSIF-2 aerodynamic system require a complete seal of the mask rim to the face. Literature frequently cites mask rim leaks as causing flow measure inaccuracies, but quantitative studies of inaccuracies are needed. Prior work (May & Scherer, in press) provided a general empirical equation relating mask rim leak flow, the cross sectional area of the rim leak, and upstream airflow. The current empirical bench research extends this DC flow work to an AC flow situation as in the case of vibrato and/or tremor using 3 and 6 Hz, 2 leak areas, 50 and 200 cm^3/s peak-to-peak variations, and an upstream flow of 200 cm^3/s. Results: smaller leak area, higher AC frequency, and higher AC airflow extent resulted in the lowest mask rim leak airflow.

3pSC12. A method for quantifying vowel change directionality. Miranda R. Morris (Linguist, UC Davis, 1 Shields Ave., Davis, CA 95616, mirmor@ucdavis.edu)

Vowels are a common variable used in studies on phonetic accommodation and imitation. The standard practice for measuring changes in vowels provides an absolute value of the distance between a baseline vowel production and a target relative to the distance between shifted/imitated vowels and that same target. While distance alone is useful, a vector based approach would form a more complete picture as capturing distance alone can mask phenomena such as exaggeration. Vowels are a two-dimensional variable; thus, a directionality measure for changes in vowels relative to some target is crucial for fully capturing phenomena in imitation. Proposed is a trigonometric approach to quantifying the directionality of vowel changes. While this method was created to aid in studies on phonetic imitation, it is also potentially useful for any work involving vowel change with a known acoustic target.

3pSC13. Evaluating the LENA recording system for investigating speech input in a French-English bilingual context. Adriel John Orena (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8th Fl., Montréal, QC H3A 1G1, Canada, adriel.orena@mail.mcgill.ca), Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada), and Julia Srouji (Dept. of Psych., McGill Univ., Montréal, QC, Canada)

The Language ENvironment Analysis (LENA) recording system is rapidly gaining acceptance as a useful tool for assessing a child's speech input. The success of this tool is due to LENA algorithms which provide automated measures of the child's auditory environment, including the amount of words. The adult word count (AWC) measure has been validated in several monolingual language contexts, including English, French, Dutch, and Mandarin. Here, we evaluate LENA performance in counting words when two languages are present within the child's input stream. Twenty French-English bilingual families with a 10-month-old infant contributed full-day recordings. Nine 5-minute segments from each family was transcribed, for a total of 900 minutes of transcribed speech. We assessed the accuracy of LENA-generated AWC for French-English bilingual speech by comparing them to human-transcribed word counts. Linear mixed modelling reveals a positive linear relationship between these two measures. Critically, the correlations between these two measures were strong for both French and English, and these correlations were not significantly different from one another. These results confirm that the LENA algorithms are sufficiently accurate in counting words in two languages, and provide support for researchers and clinicians wishing to use the LENA recording system to assess bilingual speech.

WEDNESDAY AFTERNOON, 9 MAY 2018 GREENWAY H/I, 1:00 P.M. TO 3:00 P.M.

Session 3pSP

Signal Processing in Acoustics and Underwater Acoustics: Co-Prime Arrays and Other Sparse Arrays II

R. Lee Culver, Cochair
ARL, Penn State University, PO Box 30, State College, PA 16804

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

Invited Papers

1:00

3pSP1. Biomedical applications of sparse hemispherical ultrasound arrays. Ryan M. Jones, Meaghan A. O'Reilly (Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., Focused Ultrasound Lab (C713), Toronto, ON M4N 3M5, Canada, rmjones@sri.utoronto.ca), Lulu Deng (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), and Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Sparse arrays have been employed in biomedical ultrasound for diagnostic and therapeutic purposes to reduce the technical complexities associated with dense arrays while maintaining adequate performance characteristics. Sparse arrays are particularly attractive when large apertures are desired, which is often the case for therapy to achieve high acoustic gains and tight focal volumes. Large aperture arrays provide similar benefits on receive during multichannel beamforming of passively detected acoustic emissions, which is a promising approach for guiding cavitation-mediated ultrasound therapies. Our group has designed and fabricated sparse hemispherical phased arrays for ultrasound brain imaging and therapy. Ultrasound field simulations were carried out to optimize the spatial distribution of array elements. We have employed these devices for skull computed tomography-array registration, 3D spatial mapping of microbubble activity *in vivo* through *ex-vivo* human skullcaps via noninvasive aberration correction methods, and have harnessed this spatiotemporal cavitation information to calibrate exposure levels for safe, volumetric ultrasound-induced blood-brain barrier opening. In conjunction with techniques borrowed from optical microscopy, we have demonstrated trans-skull 3D microbubble imaging beyond the diffraction limit, which may have application in ultrasound-based cerebral angiography. This talk will review our progress to date in the development and biomedical application of sparse hemispherical ultrasound arrays.

3p WED. PM

1:20

3pSP2. Passive acoustic mapping and B-mode ultrasound imaging utilizing compressed sensing for real-time monitoring of cavitation-enhanced drug delivery. Calum Crake, Seán Finn, Laurent Marsac (OxSonics Ltd., OxSonics Ltd., The Magdalen Ctr., Robert Robinson Ave., Oxford OX4 4GA, United Kingdom, calum.crake@gmail.com), Michael Gray, Robert Carlisle, Constantin Coussios (Univ. of Oxford, Oxford, United Kingdom), and Christian Coviello (OxSonics Ltd., Oxford, United Kingdom)

Ultrasound imaging presents a high-speed, low-cost approach for monitoring of focused ultrasound (FUS) therapy, and includes both conventional (B-mode) sonography and passive acoustic mapping (PAM) of acoustic emissions (Gyöngy *et al.*, 2010, Salgaonkar *et al.*, 2009). Incorporation of novel algorithms and other signal processing techniques have improved the resolution and processing speed of PAM. However, while hardware developments such as increasing channel counts provide unprecedented data capture ability, real-time processing of the growing data stream presents an evolving challenge. Previous work has employed sparse array processing techniques for PAM including matching and basis pursuit (Gyöngy & Coviello, 2011) and co-array processing (Coviello et al, 2012). Here we propose to extend PAM utilizing compressed sensing (CS). Acoustic emissions from FUS may be sparse in several domains, e.g. due to limited regions of space and time in which cavitation is likely from a focused transducer, and correlation between sensors. By exploiting this sparsity, CS allows recovery of signals sampled well below the Nyquist limit. CS-PAM thus facilitates PAM with improved spatial resolution compared to conventional methods, from fewer measurements, allowing improved image quality and reduced computational load. The technique was demonstrated for monitoring of FUS-mediated drug delivery in a murine tumor model.

1:40

3pSP3. Separation of closely-spaced acoustics sources in an under-determined system with convex optimization. Tongyang Shi and J. S. Bolton (Mechanical Eng., Purdue Univ., Ray W. Herrick Labs., 177 S Russell St., West Lafayette, IN 47907, shi247@purdue.edu)

The ability to identify acoustical source locations accurately is critical when performing sound field reconstructions. In previous work, a monopole-based, iterative equivalent source method, wideband acoustical holography, has proven able to provide accurate noise source location in complex machines even when the number of measurements was far smaller than the number of parameters that needed to be determined in the model (see: Tongyang Shi, Yangfan Liu and J. Stuart Bolton, “Diesel engine noise source visualization with wideband acoustical holography,” SAE paper 2017-01-1874). However, at some stage, when acoustical sources are too closely spaced, the current algorithm has difficulty in separating them. This is true, especially at low frequencies, in which case current holography methods tend to visualize two closely-spaced acoustics sources as a single large source, thus losing the accuracy of the source location. In the present work, the equivalent sources were still modeled as an array of monopoles, but the source strength parameter estimation and regularization problem was formulated as a convex function and turned into an iterative convex optimization problem that can be solved by an open source code. It will be demonstrated that the new procedure is capable of reconstructing very closely-spaced acoustics sources.

2:00

3pSP4. Highly directional pressure sensing using phase gradients. Joseph S. Lawrence, Kent L. Gee, Tracianne B. Neilsen, and Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, joseph-lawrence@hotmail.com)

The recently developed Phase and Amplitude Gradient Estimator (PAGE) method [Thomas *et al.*, J. Acoust. Soc. Am. **137**, 3366–3376 (2015)] estimates active intensity with a much greater bandwidth of accuracy than traditional intensity estimates. In this formulation, the intensity direction comes from the phase gradient, which when estimated from unwrapped transfer function phases can be accurate above the spatial Nyquist frequency. The phase gradient term can also be used to express pressure as a sum of directional components, which is derived by using the continuity equation with Euler’s equation. These directional pressure components have a cosine-squared directivity, which can be increased to higher orders by raising the power of the phase gradient. This method allows for the creation of highly directional pressure sensors in one or multiple dimensions using a limited number of microphones. [Work supported by NSF.]

2:20

3pSP5. Azimuth-elevation direction finding, using three uni-axial velocity sensors as an arbitrarily sparse linear array. Yang Song, Kainam T. Wong, Salman Khan, and Mohammad Asaduzzaman Khan (Dept. of Electron. and Information Eng., The Hong Kong Polytechnic Univ., CD515, Hung Hom, Kowloon, Hong Kong., Hung Hom KLN, Hong Kong, salman.khan@connect.polyu.hk)

A tri-axial velocity sensor measures the acoustic particle-velocity vector, by all three of its Cartesian components. This particle-velocity vector equals the spatial gradient of the acoustic pressure field. Song & Wong [1] has advanced an algorithm to estimate an incident source’s azimuth-elevation (θ, Φ) direction-of-arrival using three uni-axial velocity sensors that are orthogonally oriented among themselves and placed arbitrarily in three-dimensional space. This work will focus on a particular array geometry whereby the three uni-axial velocity sensors are placed on a straight line in three-dimensional space but the inter-element spacings may be entirely arbitrary. This work will show how to estimate a far-field emitter’s azimuth-elevation direction-of-arrival, despite the three elements’ spatial separation, and despite the separation’s arbitrary length and possible sparseness. Furthermore, the Cramer-Rao Bounds (CRB) will be presented analytically for the case of the linear array being aligned along the x-axis. This work differs from [2], which requires an additional pressure-sensor. [1] Y. Song & K. T. Wong, “Acoustic direction finding using a spatially spread tri-axial velocity sensor,” IEEE Trans. Aerosp. Electron. Syst. **51**(2), 834–842 (2015). [2] Y. Song & K. T. Wong, “Azimuth-elevation direction finding, using one four-component acoustic vector-sensor spread spatially along a straight line,” Proc. of Meetings on Acoustics—Meeting of the ASA, Pittsburgh, U.S.A., May 18–22, 2015.

2:40

3pSP6. Identification of fin whale 20-Hz pulses and other vocalizations from passive monitoring with a coherent hydrophone array. Wei Huang (Elec. and Comput. Eng., Northeastern Univ., 500 Broadway, Apt 3157, Malden, MA 02148, huang.wei1@husky.neu.edu), Heriberto A. Garcia (Elec. and Comput. Eng., Northeastern Univ., Arlington, MA), and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

A large variety of sound sources in the ocean, including biological, geophysical and man-made activities can be simultaneously monitored over instantaneous continental-shelf scale regions via the passive ocean acoustic waveguide remote sensing (POAWRS) technique by employing a large-aperture densely-sampled coherent hydrophone array. Millions of acoustic signals received on the POAWRS system per day can make it challenging to identify individual sound sources. An automated classification system is necessary to enable sound sources to be recognized. Here each detected acoustic signal is represented by an amplitude weighted pitch track which describes its fundamental frequency-time and amplitude variation. Multiple features are extracted from the pitch track including mean, minimum and maximum frequencies, bandwidth, signal duration, frequency-time slope and curvature, as well as several amplitude weighted features. A large training data set of fin whale 20-Hz pulses and other vocalizations are gathered after manual inspection and labeling. Next, multiple classifiers including logistic regression and decision tree are built and tested for identifying the fin whale 20-Hz pulses and other vocalizations from the enormous amounts of acoustic signals detected per day. The performance of the classifiers are evaluated and compared

WEDNESDAY AFTERNOON, 9 MAY 2018 GREENWAY B, 1:00 P.M. TO 2:05 P.M.

Session 3pUW

Underwater Acoustics: Target Scattering in Underwater Acoustics: Imaging, Spectral Domain, and Other Representations II

Daniel Plotnick, Cochair
Applied Physics Lab., Univ. of Washington, Seattle, WA 98105

Timothy Marston, Cochair
APL-UW, 1013 NE 40th Street, Seattle, WA 98105

Chair's Introduction—1:00

Contributed Papers

1:05

3pUW1. Shallow water multichannel sub-bottom profiling using Kirchhoff imaging. João P. Ristow, Marina M. Bousfield, Gregório G. Azevedo (Mech. Eng., Federal Univ. of Santa Catarina, Rua Lauro Linhares, 657, Apto. 203B, Florianópolis, Santa Catarina 88036-001, Brazil, jpristow@gmail.com), Julio A. Cordioli (Mech. Eng., Federal Univ. of Santa Catarina, Florianópolis, SC, Brazil), and Stephan Paul (Mech. Eng., Federal Univ. of Santa Catarina, Joinville, Brazil)

Single-channel sub-bottom profilers are commonly used for shallow water seismic investigation. This work reports the design and construction of a multiple channel profiler that allows more effective seismic imaging techniques. The system is composed of a controlled acoustic source and a 1.6 m long array with eight evenly spaced receiver channels, and operates using CHIRP signals with frequency ranging from 800 Hz to 8 kHz. A survey was carried out to obtain seismic data in the region of Lagoa da Conceição, Santa Catarina, Brazil. The acquired data was processed using a seismic imaging technique based on the pre-stack Kirchhoff time migration. Both the multichannel seismic data acquisition system and the applied imaging technique showed to be robust for acquiring subsurface information. The use of the pre-stack Kirchhoff time migration resulted in a seismic image with reflection events allocated in the real position and without bowtie effects since it repositions the diffracted energy back to the spreaders that originated it. This procedure results in a more reliable subsurface image of the studied environment, when compared with single-channel sub-bottom profilers.

1:20

3pUW2. Effect of mesoscale eddies on deep-water sound propagation. Yao Xiao (Inst. of Acoust., Chinese Acad. of Sci., 801 Ferst Dr., George W. Woodruff School of Mech. Eng., Atlanta, GA 30332, xyao62@gmail.com), Zhenglin Li (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), and Karim G. Sabra (Mech. Eng., Georgia Tech, Atlanta, GA)

The mesoscale eddies are the most common mesoscale phenomena in the ocean. The cold and warm water carried by the mesoscale eddies can significantly change the structure of the sound speed field, which is the key factor affecting long range deep-water sound propagation. In this presentation, a deep-sea analytical eddy model [Henrick *et al.*, JASA **62**, 860–870(1977)] is used to determine the sound speed distributions produced by cold eddy in the southwest of the South China Sea. The statistical characteristics of the influence of eddies on the acoustic propagation are investigated with a range-dependent parabolic equation (PE) model to asses the effects of the relative position of sound sources and eddies, the frequency sensitivity, the acoustic energy deviation of the convergence zone and the shadow zone for long-range (600 km) low frequency (50 Hz) sound propagation in

this ocean model. Specific features of the induced sound propagation variability such as the travel time deviation and the standard deviation of ensemble intensity Will be quantified. Finally, the refraction due to mesoscale eddies on large scale low-frequency propagation is discussed using the full three-dimensional parabolic (FOR3D) and explaining by the full three-dimensional ray theory (Bellhop3D).

1:35

3pUW3. Scattering of low-frequency sound by infinite fluid and solid cylinders. Alexander B. Baynes and Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, Spanagel Hall, Rm. 215, Monterey, CA 93943-5216, abbaynes1@nps.edu)

Wave scattering by obstacles is typically studied assuming plane wave incidence. However, full Green's functions are necessary for problems where the separation of the scatterer from sources, interfaces, or other scatterers is comparable to the dimensions of the scatterer itself. In this paper, the two-dimensional problem of scattering of monochromatic cylindrical waves due to infinite cylinder embedded in a homogeneous fluid is considered. Fluid and solid cylinders are studied, and soft and hard cylinders are revisited. The exact solutions for the Green's functions are expressed as an infinite series of cylindrical functions with complex amplitudes determined by the acoustic boundary conditions at the surface of the cylinder. Here, we derive closed-form asymptotics for the scattered field in the regime when the scatterer dimensions are small compared to wavelength, i.e., Rayleigh scattering. The scattered wave approximation is valid for arbitrary source and observation points outside the scatterer and is expressed as a simple sum of fields due to three image sources. Image source solutions were anticipated due to classically studied electrostatic analog problems involving dielectric cylinders. Image source representation allows physical insight into the scattering physics and suggests simple yet accurate analytic solutions to interface and waveguide scattering problems.

1:50

3pUW4. Scattering of low-frequency sound by shallow underwater targets. Alexander B. Baynes and Oleg A. Godin (Dept. of Phys., Naval Postgrad. School, Spanagel Hall, Rm. 215, Monterey, CA 93943-5216, abbaynes1@nps.edu)

Rayleigh scattering of sound by a target can be described as a wave radiated by virtual point sources inside the target. When a target is located close to the ocean surface or another reflecting boundary, reflections of the incident and single-scattered waves from the boundary lead to multiple scattering from the target, with the target being insonified by nearby virtual sources. At low frequencies and for shallow targets, the distance from a virtual source to the target is not necessarily large compared to acoustic wavelength or the target's dimensions. This paper takes advantage of the virtual source concept and recently derived explicit analytic representations of 2-D and 3-D acoustic Green's functions in unbounded fluids with inclusions of a circular cross-section, to develop a simple and numerically efficient model of multiple scattering. Scattering from soft, hard, fluid, and solid objects is considered. The model is used to study the acoustic field in the vicinity of shallow targets, calculate normal-mode compositions of the multiple-scattered field, examine conditions of applicability of the single-scattering approximation, and investigate implications of multiple scattering for target detection and classification.

Plenary Session and Awards Ceremony

Marcia J. Isakson,
President, Acoustical Society of America

Annual Membership Meeting

Presentation of Certificates to New Fellows

Deniz Baskent – For contributions to our understanding of acoustic and electric auditory and speech perception

Monita Chatterjee – For contributions to cochlear implant psychophysics and speech perception

Patricia Davies – For contributions to the fields of sound quality, aircraft noise, and the dynamic properties of foams

Jin-Yong Jeon – For contributions to research and practical design in soundscape and architectural acoustics

Bruce C. Olson – For contributions to design and modeling of electro-acoustic systems for architectural spaces

Lauri Savioja – For contributions to room-acoustics modeling and auralization

Christopher J. Struck – For contributions to acoustical standards development and instrumentation design

Introduction of Award Recipients and Presentation of Awards

William and Christine Hartmann Prize in Auditory Neuroscience to Shihab Shamma

Medwin Prize in Acoustical Oceanography to Yin-Tsong Lin

R. Bruce Lindsay Award to Yun Jing

von Békésy Medal to David Kemp

Helmholtz-Rayleigh Interdisciplinary Silver Medal to Kenneth S. Suslick

Gold Medal to William A. Yost

Vice President's Gavel to Michael J. Buckingham

President's Tuning Fork to Marcia J. Isakson

Session 3eED

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

Peggy B. Nelson, Cochair
Univ. of Minnesota, Minneapolis, MN 55455

Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, NY 11787

This workshop for Minneapolis area Girl Scouts 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.

WEDNESDAY EVENING, 9 MAY 2018 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 hold its meeting 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 Wednesday are as follows:

Biomedical Acoustics	Greenway F/G
Signal Processing in Acoustics	Greenway H/I

ACOUSTICAL SOCIETY OF AMERICA

R. BRUCE LINDSAY AWARD

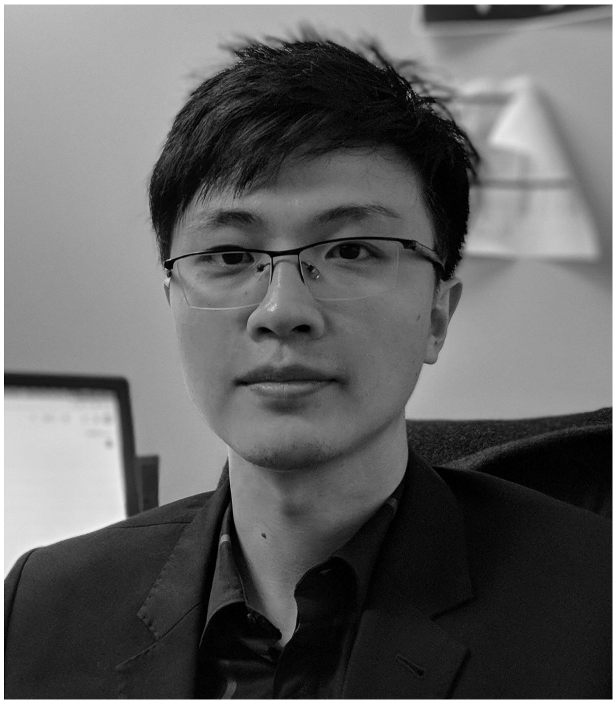
Yun Jing

2018

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



CITATION FOR YUN JING

. . . for contributions to acoustic metamaterials and numerical modeling of wave propagation in rooms and complex media

MINNEAPOLIS, MINNESOTA • 9 MAY 2018

Yun Jing was born into an intellectual family in Nanjing, China. His maternal grandfather was a renowned Chinese scholar of linguistics, his mother is a novelist, and his father is a professor of Chinese literature. Unlike his parents and grandfather, Yun developed an interest in engineering rather than in literature. Yun chose acoustics as his undergraduate major and graduated in 2006. He earned M.S. and Ph.D. degrees in 2007 and 2009, from the Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute (RPI). He is currently a tenured Associate Professor in the Department of Mechanical Engineering, North Carolina State University (NCSU). Before joining the faculty at the NCSU, Yun also spent two years from 2009 to 2011 as a Research Fellow at Brigham and Women's Hospital at Harvard Medical School.

A few weeks after arriving at RPI in the fall of 2006, Yun asked his advisor for possible directions for his graduate research. When he was given a paper on experimental investigations using 3-D p-u sensors that was just published in the *Journal of the Acoustical Society of America* (JASA), Yun responded with moderate enthusiasm and asked for other options. Two days after he was given a more mathematical and theoretical JASA publication on diffusion theory in room acoustics, he wrote to his advisor that, "not only do I understand the underlying theory, but I also implemented the work in this paper and came up with exactly the same results as published." Two months later he demonstrated in a draft JASA letter that the current application of the diffusion equation in room acoustics can produce improved results using an Eyring boundary condition [Jing and Xiang, JASA **121**, 3284–3287, 2007]. He soon incorporated more rigorous boundary conditions into diffusion equation models, which led to his second peer-reviewed journal publication [Jing and Xiang, JASA **123**, 145–153, 2008], 10 months after his arrival at RPI. His work on the boundary conditions allows rigorous application of diffusion theory for efficient room acoustic predictions of sound energy flow and decay in enclosed spaces. Yun's accomplishments resulted in the successful completion of his M.S. degree in Architectural Acoustics, with two JASA publications in less than one year.

The transport theory has drawn attention in the urban noise community in the context of modeling street canyons since about 2004. However, no solution of the transport equation was available then that was useful for this application. Within the scope of his doctoral research, Yun developed a collaboration with Prof. Ed Larsen of the Department of Nuclear Engineering and Radiological Sciences at the University of Michigan. As a result, his thesis contains the methodology for solving the transport equation to numerically model room acoustics energy propagation, along with experimental validation of the theory implemented for long-space configurations. Thus, Yun produced pioneering work on transport equation-based room acoustic modeling, successfully finishing his Ph.D. dissertation at RPI in August, 2009, in less than two years. He and his advisors published seven JASA papers along this line of research.

Yun subsequently went further to pursue postdoctoral research at the Brigham and Women's Hospital of Harvard Medical School. He worked under Dr. Greg Clement, studying extensions of the angular spectrum approach for modeling of nonlinear ultrasonic wave propagation. This was a major shift in research direction from anything Yun had done in architectural acoustics. Nonetheless, Yun was never one to shy away from taking on new challenges such as understanding nonlinear acoustics, with gratitude to Hamilton and Blackstock's book. Yun and Greg discovered that the Westervelt equation in the wavenumber-frequency-domain, can be solved using a one-dimensional Green's function. This allowed them to propose and investigate a modified angular spectrum approach. Following his initial ultrasonic work, Yun became intrigued by numerical modeling of ultrasound wave propagation, leading him to publish several other journal papers on this topic. For example, he developed another algorithm for solving the Westervelt equation based on the wavenumber-space method. It has the advantage that medium heterogeneities can be taken into account to model more realistic biological tissue. Subsequently, Yun used this method for finding the phase correction of transcranial beam focusing, which he continues to work on because of its usefulness in medical ultrasound.

At NCSU Yun actively pursues work in architectural acoustics and numerical modeling of ultrasound wave propagation. In addition, he has branched out into two different areas, acoustic metamaterials and therapeutic ultrasound. Yun realized the excitement and rapid growth in the field of metamaterials, which provided him with many opportunities to explore. His group recently published 13 journal articles on this topic, including four in two of the most prestigious physics journals, *Physical Review Letters* (PRL) and *Physical Review X*. One of his PRL papers was selected as an "Editor's Suggestion" [*PRL* **115**, 254301, 2015]. In this paper, Yun's group proposed a simple structure consisting of paper and aluminum that exhibits a negative effective density in one direction and positive effective density in another, giving rise to a peculiar hyperbolic dispersion. Such a hyperbolic metamaterial can be used for applications such as energy focusing, tunneling, and subwavelength imaging. This is the first time that a broad-band, truly hyperbolic acoustic metamaterial was proposed and experimentally validated.

Yun plays an active role in the affairs of the Society. In addition to attendance with paper presentations at meetings of the Acoustical Society of America (ASA) since 2006, he has been invited many times to deliver invited papers at ASA meetings and other conferences, he has recently organized and chaired three special sessions and chaired a number of other sessions during the ASA meetings.

Dr. Yun Jing is an exceptionally creative young scientist who has demonstrated a track record of high creativity and self-direction. Yun has made an important impact on the state of knowledge regarding numerical modeling of wave propagation in complex media and acoustic metamaterials. We are delighted to see the Acoustical Society of America recognizing and honoring Dr. Yun Jing with its prestigious R. Bruce Lindsay Award.

NING XIANG
WILLIAM SIEGMANN

ACOUSTICAL SOCIETY OF AMERICA

VON BÉKÉSY MEDAL

David Kemp

2018

The von Békésy Medal is presented to individuals, irrespective of nationality, age, or society affiliation, who have made outstanding contributions to the area of psychological or physiological acoustics as evidenced by publication of research results in professional journals or by other accomplishments in the fields.

PREVIOUS RECIPIENTS

Jozef J. Zwislocki	1985
Peter Dallos	1995
Murray B. Sachs	1998
William S. Rhode	2010
M. Charles Liberman	2012

CITATION FOR DAVID T. KEMP

. . . for the discovery of otoacoustic emissions and contributions to cochlear biophysics and the detection of hearing loss

MINNEAPOLIS, MINNESOTA • 9 MAY 2018

David Kemp trained as a geophysicist, earning his Ph.D. in physics from the University of London in 1970. His early research focused on the prominent series of peaks in the extreme low-frequency (ELF) region of the Earth's electromagnetic spectrum known as "Schumann resonances." Schumann resonances are global electromagnetic standing waves, excited by lightning discharges, that form in the "cavity" bounded by the Earth's surface and the ionosphere. With this background, David was well poised to grasp the significance of the curious, quasi-periodic pattern of spectral peaks and valleys ("microstructure") evident in the hearing threshold curve. David immediately understood this microstructure as the tell-tale signature of standing waves inside the "cavity" of the cochlea. These standing waves, he realized, should be detectable outside the cavity, in the external ear canal, if only one could listen hard enough. And thus, by listening to the ear, David's soon-to-be-illustrious career in auditory neuroscience was born.

By having the genius to do the simple and obvious—placing a miniature low-noise microphone in the human ear canal and carefully recording and interpreting the ear's response to sound—David Kemp sparked nothing short of a revolution in our understanding of the physical and physiological basis of hearing. David's discovery in the mid-1970s that the ear makes sound while listening to sound completely overturned the reigning dogma of the day that the cochlea was nothing but a linear, passive mechanical transducer.

Early on, many refused to believe it, regarding the "Kemp effect" as some sort of insidious artifact. But these critics overlooked David's brilliant series of well-controlled experiments establishing beyond doubt that the sounds he was recording originated within the inner ear. Others, although willing to credit the result, failed to appreciate the extraordinary theoretical and practical importance of these sounds, which are now known as otoacoustic emissions. For this reason, David experienced difficulty getting his results published. The manuscript announcing the discovery was rejected by *Nature* on the grounds that otoacoustic emissions would surely prove of little interest outside the community of clinicians concerned with the diagnosis of hearing impairment. Thus, thanks to the obtuseness of *Nature*'s editors, the Acoustical Society of America (ASA) received the honor of publishing David's seminal results. David's initial report is now the fourth most-cited paper in the history of the *Journal of the Acoustical Society of America* (and, according to the Science Citation Index, is the only one in the top nine concerned with auditory physiology).

David's discovery and his subsequent studies probing the physical and physiological mechanisms confirmed controversial hints of cochlear mechanical nonlinearity and revived a largely forgotten suggestion—by astrophysicist Thomas Gold in 1948—that the cochlea employs "regenerative action" powered by electrochemical potentials to enhance cochlear frequency selectivity by counteracting the viscous damping. Thus, David's work launched a flurry of experimental and theoretical activity, and within just a few short years the paradigm had shifted completely—the inner ear was now no mere passive transducer, but an active nonlinear amplifier controlled via neural feedback from the brain. Researchers soon found that the cochlea contains an array of cellular force generators (outer hair cells) that act in concert to amplify quiet sounds, thereby boosting the sensitivity and dynamic range of hearing.

Today, the quest to understand the mechanical, cellular, and molecular basis of the "cochlear amplifier" and its feedback control mechanisms dominates the study of the peripheral auditory system, and understanding the role of nonlinear amplification and compression in the analysis and perception of sound is a central theme in psychophysics. In the years since Kemp's discovery, the revolution has spread throughout the animal kingdom. Otoacoustic emissions (OAEs) almost identical to those first discovered by Kemp have been reported in animals as diverse as birds, lizards, frogs, and insects. The biological amplification of sound appears to be a universal theme in sensory physiology.

All of this is the direct result of David's singular contribution to basic science. But David had the further insight that OAEs had enormous translational potential. He realized that the emitted sounds provide information about the health of the inner ear—because changes in emission magnitudes reflect changes in hearing sensitivity, OAEs can be used as an objective test for the presence of hearing impairment. David showed that OAEs could be recorded using a small probe—containing both an earphone to provide the stimulus sound and a microphone to detect the emission—inserted into the ear canal. Recording OAEs was therefore a simple, noninvasive procedure capable of detecting sound-evoked biological activity in the inner ear.

After validating this procedure as an objective test of auditory function—and having failed to obtain the necessary support, either from the hearing test industry or from government bodies—David established his own company, Otodynamics Ltd, to develop, manufacture, and refine the test equipment and its software. With the unwavering support of his wife, Gillian, David built the company while continuing his full-time academic commitments as a lecturer and then Professor of Auditory Biophysics at University College, London (UCL). The resulting "ILO88" instrument set the standard for what has become a vital piece of every audiologist's armamentarium. Today, OAE recording is an indispensable component of the battery of audiological tests used in hospitals and clinics around the world; literally thousands of ILO88s and its successors are testing hearing every day, in both advanced and developing countries. OAE recording has also become a routine procedure in basic science laboratories investigating the mechanisms underlying normal hearing and its pathologies.

As noninvasive tests that require no input from the patient, OAE measurements are particularly valuable for assessing the hearing of babies. David recognized this potential and saw the value of early hearing screening so that intervention for hearing impairment might be implemented early in life. He was at the forefront of the drive for screening all neonates for hearing impairment. Newborn screening was first implemented in the USA, later in Europe and the UK. The early identification of hearing impairment is, of course, of enormous significance for the social and educational development of affected children.

David has used resources from his company to help build a premier auditory research facility, the Centre for Auditory Research at the UCL Ear Institute. This facility now houses many of the world's top researchers devoted to the study of hearing in all its aspects, including auditory biophysics, psychophysics, neuro- and molecular biology, and genetics. Academically, David has led the development and teaching of undergraduate degree courses in Audiology and postgraduate degree programs in Audiological Science and Audiological Medicine, as well as training and mentoring the next generation of researchers in the field though his supervision of Ph.D. students and postdoctoral fellows. At hearing conferences, David is often sighted quietly haunting the poster aisles, patiently probing and encouraging those young researchers now following in his footsteps.

Over the years, David has received many honors and awards recognizing his outstanding contributions to basic and applied hearing science. These include the Association for Research in Otolaryngology's Award of Merit, the American Speech-Language-Hearing Association's Distinguished Service Award, and his election as a Fellow of the Royal Society. David is a supreme example of how science, technology, and innovation can combine for the betterment of society. He has played a leading role in developing the measurement and analysis of OAEs into a ubiquitous and invaluable tool for improving human health. We are very pleased to congratulate David Kemp for being awarded the Acoustical Society of America's von Békésy Medal.

CHRISTOPHER A. SHERA
ANDREW FORGE
BRENDA L. LONSBURY-MARTIN

ACOUSTICAL SOCIETY OF AMERICA
HELMHOLTZ-RAYLEIGH INTERDISCIPLINARY
SILVER MEDAL
in

Physical Acoustics and Biomedical Acoustics

Kenneth S. Suslick

2018

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.

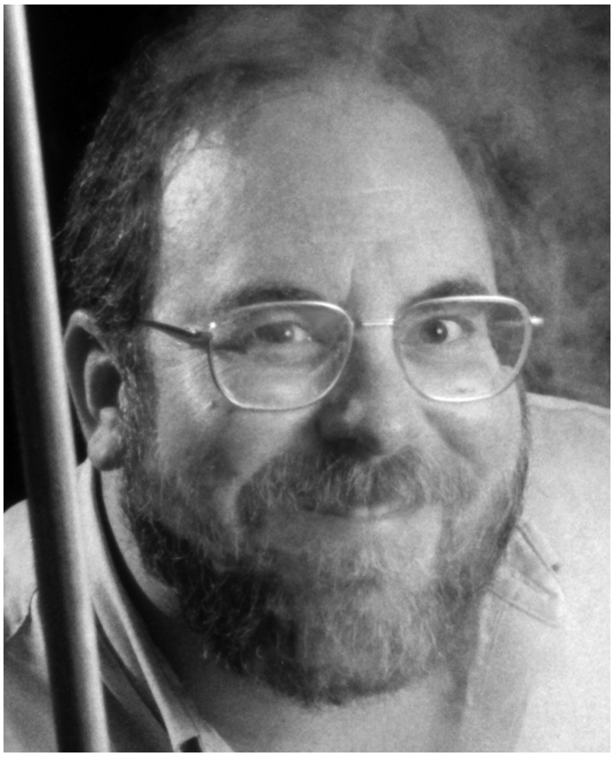
PREVIOUS RECIPIENTS

Helmholtz-Rayleigh Interdisciplinary Silver Medal

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

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 KENNETH S. SUSLICK

. . . for contributions to the acoustics of sonochemistry

MINNEAPOLIS, MINNESOTA • 9 MAY 2018

Ken Suslick received his B.S. from the California Institute of Technology in 1974, his Ph.D. from Stanford University in 1978, and moved immediately to the University of Illinois where he is now Marvin T. Schmidt Research Professor of Chemistry. A prolific scientist, he has published nearly 400 papers, edited four books, and holds 48 patents and patent applications. As a modern measure of scientific productivity, he is listed under Google Scholar as having over 46,000 citations and an h-index of 109—which means 109 of his papers have been cited at least 109 times.

Among his numerous other research recognitions, He is the winner of many awards, among them: Fellow, National Academy of Inventors; Centenary Prize, Royal Society of Chemistry; Sir George Stokes Medal, Royal Society of Chemistry; Materials Research Society Medal; Silver Medal of the Royal Society for Arts, Manufactures, and Commerce; the American Chemical Society Nobel Laureate Signature Award, and notably, the Acoustical Society of America's Student Council Mentor Award. In 2018, Professor Suslick has been selected as the 76th George Eastman Professor of Oxford University, which is a distinguished Chair appointed annually to a citizen of the United States who is deemed to be of great eminence in teaching or research in any field of study at the University of Oxford. Prior Eastman Professors include 12 Nobel Prize winners.

In addition to his academic research, Professor Suslick has had significant entrepreneurial experience. He was the lead consultant for Molecular Biosystems Inc. and part of the team that commercialized the first echo contrast agent for medical sonography, Albunex™, which became Optison™ by GE Healthcare. These ultrasound contrast agents are, stabilized microbubbles, which have proved to be a major recent contribution to medicine, as the huge scattering cross section of a bubble in resonance permits diagnostic ultrasound scanners to see blood vessels down to the capillary level. In 1990, he invented core-shell microspheres with albumin outer coating and hydrophobic cores using ultrasound for both emulsification and chemical crosslinking of the protein shell. This lead to his role as the founding consultant for VivoRx Pharmaceuticals and the commercialization of Abraxane™, albumin microspheres with a paclitaxel core, which is the predominant delivery system for taxol chemotherapy for breast cancer; VivoRx became Abraxis Bioscience, which was acquired by Celgene for $2.9 billion. He also co-founded ChemSensing and its successors, iSense Systems and Specific Technologies in Mountain View, for the commercialization of optoelectronic nose technology with particular focus on biomedical applications of this unique sensor technology.

Ken has practically invented an entire field of science, namely, that of "sonochemistry." When a liquid, say, water is subjected to an acoustic field of sufficient intensity; the liquid literally tears apart during the rarefaction cycle. The cavity that is formed rapidly grows, filling with vapor, until the compression cycle, and since the vapor can't supply any resistance beyond its vapor pressure, the cavity collapses. When the cavity collapses, any remaining non-condensable gas is subjected to enormous pressures and temperatures. This process is called acoustic cavitation and is the principal basis of sonochemistry. Ken's papers in the early 1980's were instrumental in opening the diversity of cavitation driven chemistry. Sonochemistry as a field has expanded greatly over the past twenty years with a dedicated journal (of which Ken was a co-founding editor) that publishes hundreds of papers annually. One of the consequences of cavitation collapse is that the gas that is compressed is heated to incandescent temperatures—often higher than that of the surface of the sun. This phenomenon is responsible for sonochemical reactions, and also for the emission of light, called sonoluminescence. Ken's articles on sonoluminescence have made the covers of *Science* (twice) and of *Nature*. At one time, over 1,000 papers were published on the topic in a single year, and Hollywood even made a movie (*Chain Reaction*) about it starring

Morgan Freeman and Keanu Reeves, to which Ken and his grad students served as uncredited consultants.

Ken's ingenuity was displayed by his use of cavitation to create completely unique materials. If cavitation is performed in a volatile metallic liquid, the high temperature within the compressed bubble breaks up the molecules into their constituent atoms, and as the bubble cools, the atoms can condense into nanoparticles. Since the cooling rates are greater than 10^{10} Kelvins/second, unique particles are formed, such as amorphous iron or novel metal alloys—which can't be created using conventional means. For this work, and similar research, he was awarded the Materials Research Society Medal.

Ken is not only a brilliant scientist but also an amateur artist and an art collector. Having a special interest in the human face, Ken has created numerous bronze casts of abstracted heads (in the art foundry at the University of Illinois) that display his unique creativity. Of special interest to Ken are ethnographic masks, with a collection of over 400 from all over the world, displayed on the walls of their house and his office: his wife says that there is hardly any wall space left! Ken's mother was an artist and Ken collected his first mask when he was eight, and he still has it. Ken is also known for his memory of lyrics from folk music, Tom Lehrer, and Gilbert & Sullivan (as well as bawdy songs and ballads) and has been an active host to house concerts by musicians passing through central Illinois.

Ken was born in Chicago, eldest of four children of Alvin (M.D.) and Edee (R.N.). He grew up in Glencoe, IL, a suburb on the north shore of Chicago. Ken's son, Benjamin, is also a Caltech alum and is currently pursuing his Ph.D. (in chemistry) at U.C. Berkeley. Ken's wife, Patricia, is a retired school librarian and has previously taught at levels ranging from elementary school up to college freshmen. They have two cats, one nice to Ken and one not so nice.

LAWRENCE A. CRUM
ANDREA PROSPERETTI
RONALD A. ROY

ACOUSTICAL SOCIETY OF AMERICA
GOLD MEDAL

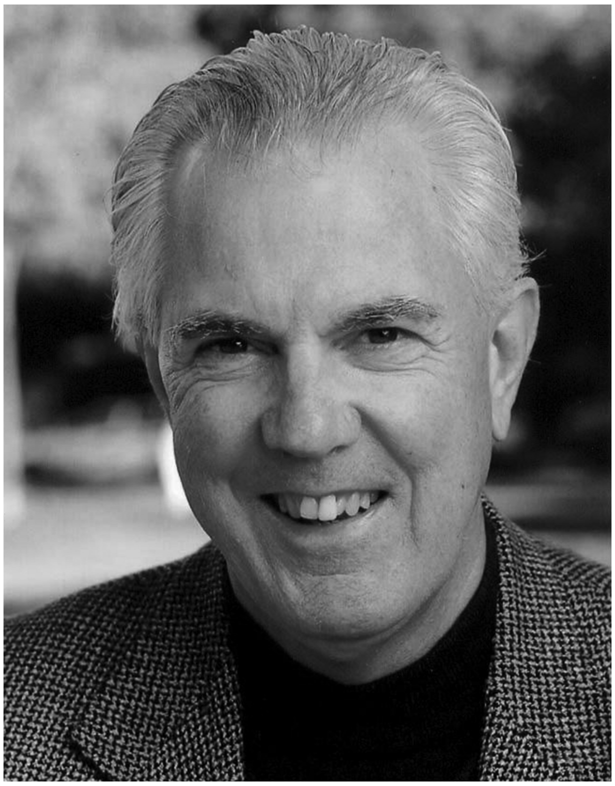
William A. Yost

2018

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



CITATION FOR WILLIAM A. YOST

. . . for research on binaural hearing, pitch and modulation perception and for service to the acoustics community

MINNEAPOLIS, MINNESOTA • 9 MAY 2018

William Albert Yost (Bill) was born in 1944 and grew up in Colorado. He earned his undergraduate degree from Colorado College (1966) where he studied psychology and mathematics and played varsity basketball and tennis. Colorado College would later recognize him with an honorary degree in 1997. The combination of psychology and mathematics took Bill to Indiana University–one of a few American centers where psychologists were transforming one corner of their field into something like a hard science through signal detection theory. During his Indiana days, Bill's romance with Lee began to bloom. They married in 1969 and have two daughters and four grandchildren. Bill received the Ph.D. in Experimental Psychology in 1970 and, like so many others of his generation, began a postdoctoral fellowship with David Green at the University of California, San Diego.

Bill's postdoc time was productive but brief. In 1971 he joined the faculty at the University of Florida–jointly in communication science and in psychology. In 1977 he moved to Loyola University of Chicago where he remained for the next 30 years. At Loyola he served as professor of hearing science, professor of psychology, and director of the Parmly Hearing Research Institute. Later he served as associate vice president for research and dean of the graduate school. In 2007, after Bill and Lee decided that they had seen enough bad weather, Bill became chair and professor of hearing science at Arizona State University where he is now Research Professor.

Throughout Bill's long and varied career, there has been one constant, namely psychoacoustics research. His CV lists more than 100 refereed publications, 67 of them in the *Journal of the Acoustical Society of America* (JASA)... and counting. Bill's training as a student and postdoc prepared him for research in theory and experiment, especially as applied to binaural hearing. He thoroughly absorbed the signal detection techniques, as evidenced by papers from early in his career on the binaural masking level difference. Subsequently, he moved in directions of more directly measurable perceptions, namely binaural sound localization/lateralization and pitch perception.

In the binaural area, Bill did very important experiments associating interaural differences in signal arrival time and signal level with perceived lateralization within the head. Such measurements are notoriously difficult because of biases. It is much easier to do an experiment that asks the listener to identify a change in laterality than to identify laterality absolutely. Not surprisingly, his 1981 paper on the lateral position of sinusoids is his most widely cited binaural research paper. Bill also wrote, together with Ruth Litovsky, Steve Colburn, and Sandy Guzman, the widely cited JASA review of the binaural effect known as the "precedence effect." Overall, Bill's work on localization/lateralization was creative, deep, and impactful, and it, together with his work on the Franssen effect and cocktail party effect, became the cornerstone of his many talks for a general audience and popularizations. Recently, Bill has developed a novel facility to study auditory spatial perception when the source of sound is rotating, or the listener is rotating, or both are independently rotating. This fundamental research has obvious importance for our abilities to understand the world around us and may ultimately be clinically helpful for impaired observers. This extension of binaural research is also fun to think about.

Bill's contributions to the science of pitch have been on the perception of rippled noise. Rippled noise is the result of adding a delayed version of a noise to the original noise itself. The effect is not new. The pitch of rippled noise was discovered by Huygens in 1693 and modern experiments began in The Netherlands in the late 1960s. Bill entered the field in the 1970s struggling with the problem that occurs when the noise is added back with a 180-degree phase shift. That stimulus gives rise to two different, but rather similar, pitches and neither directly reflects the delay. He would go on to write five more articles on rippled noise pitch, pitch strength, and modeling. Then in 1993 Bill discovered iterated rippled noise, where the delay-and-add process for rippled noise is repeated to make finite or infinite impulse response filters. That variation led to 11 more publications, and these are Bill's most widely cited research articles.

Underlying the interest in rippled noise pitch is the hope that the combination of theory and experiments will clarify the origin of pitch itself. Thus, although Bill did not invent rippled noise, he brought the study to the United States and promoted it at every opportunity. As a result, rippled noise became a standard stimulus. Articles in JASA show it applied to hearing impairment, vertical plane localization, infant screening, auditory filter measurement, cochlear implants, chinchillas, budgerigars, bottlenose dolphins, and beluga whales.

Starting in the 1980s Bill did important early work on modulation detection interference (MDI), including the important step of adding interaural differences. MDI is observed when a slow amplitude modulation in one signal or noise (called the masker) interferes with the ability to detect amplitude modulation in a different signal or noise. The effect occurs even if the two signals are separated by several octaves. Interference is greatest when the masker has the same modulation frequency as the signal. The unimportance of carrier frequency and the tuning in modulation clearly emerged from Bill's work. The problem with MDI is to understand its origins. It is known that there are tuned modulation channels in the auditory system, and MDI may reflect nothing more than masking within these channels. Alternatively, MDI may have a central origin, reflecting the ability of common modulation to fuse signals perceptually, such as two tones in very different frequency ranges.

From the earliest days of his academic career Bill has contributed to education in hearing science. His undergraduate textbook *Fundamentals of Hearing*, originally co-authored with Donald Nielsen in 1977 and now in its 5th edition, has introduced hearing science to hundreds of thousands of students. He is the author of five other books and 45 book chapters or conference proceedings. Over the years Bill has mentored junior colleagues who remain active in research and in the ASA. Colleagues find that Bill's excellent memory for facts and ideas in psychoacoustics is a valuable resource.

Everywhere he goes Bill gets involved with what is going on. His record shows contributions to noise standards for forensic scientists, noise ordinances for Florida, chairing committees for the National Research Council, lectures on computers in education, and letters to *Science*.

Bill gave a talk at the Cleveland meeting of the ASA in November 1968, and he has been active in the Society ever since. He has held every elective office that the Society has available, serving as president in 2005. He was the ASA co-chair of the largest acoustics meeting in history (ASA-EAA Paris, 2008). Although he has been active as a member and an officer of other scientific organizations, the ASA has been his main intellectual home, and the ASA has benefited from his administrative skills. Bill is the consummate committee chair. He has an uncanny ability to read the sense of the room and to guide discussions to successful conclusions.

Because of Bill's talents as an administrator, a convener, and a leader there was really no limit to how far he might have gone in university or government administration. With his background and personality he could have assumed high profile roles elsewhere if he had wanted to do it. Fortunately for us, he remained dedicated to research in psychoacoustics, and we find it most appropriate that the Society awards him its Gold Medal.

WILLIAM M. HARTMANN
ROY D. PATTERSON
H. STEVEN COLBURN