Session 3aAA

Architectural Acoustics, Noise, ASA Committee on Standards, and Structural Acoustics and Vibration: Advances in the Laboratory Testing of Materials

Ronald Sauro, Chair
NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541

Chair's Introduction—7:55

Invited Papers

8:00

3aAA1. Common problems associated with reverberation rooms in sound testing laboratories. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Reverberation rooms have proven to be a very useful tool in acoustic laboratories for measuring the sound absorption of acoustic materials, airborne and impact sound transmission through building partitions, as well as measuring the overall radiated sound power level of a wide variety of sound sources. The primary benefit of a reverberation room is that the measurement can be made in much less time than with alternative environments (e.g., anechoic or hemi-anechoic chambers, or via acoustic intensity). The primary disadvantage of the reverberation room measurements is that all information regarding the directivity of the acoustic source is lost. This paper discusses some of the design and acoustical performance problems that are often found in reverberation rooms that could have been avoided with proper foresight. Issues that will be discussed include location relative to nearby noise and ground vibration sources, room size and shape, space planning for microphone locations, low frequency absorbers, and the importance of temperature and humidity control. Specific guidelines that should be followed when planning a new reverberation room will be provided.

8:20

3aAA2. Measuring with noise? We can do better!. Markus Müller-Trapet and Christoph Hoeller (National Res. Council Canada, 1200 Montreal Rd., Ottawa, ON K1A 0R6, Canada, Markus.Mueller-Trapet@nrc-cnrc.gc.ca)

It has long been established that correlation-based measurement techniques, using, e.g., maximum length sequences and swept sines, outperform classical methods based on random Gaussian noise. Among the advantages of these modern techniques are better rejection of background noise and reduced measurement duration to achieve similar or better precision. These advantages are especially interesting in room and building acoustics, where many measurement positions typically have to be covered and background noise is often an issue, especially when measuring sound transmission loss or velocity level differences. Unfortunately, there is currently no provision in the ASTM standards on room and building acoustics that would allow the use of these modern measurement methods. To demonstrate the advantages of these methods, this contribution will present an example of measurements of the apparent sound transmission loss, i.e., measurements of sound pressure level differences and reverberation times. In addition to the standardized measurements according to ASTM E336, all measurements were repeated with maximum length sequences and swept sines. The results will be compared and the advantages of the modern techniques will be highlighted.

8:40

3aAA3. Designing reverberation chambers for improved low frequency response. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Most reverberation rooms are able to provide good diffusion and spatial uniformity in the reverberant field of the source at mid- and high-frequencies. Meeting the broadband spatial uniformity requirements for ANSI S12.51 (ISO 3741) is relatively easy to achieve in the frequency bands 100 Hz to 10,000 Hz provided that the recommended room volumes are used. However, meeting the spatial uniformity requirements becomes much more difficult when the desired frequency range includes frequency bands below 100 Hz and the room must meet the pure tone qualification criteria (as is required in AHRI 220). There are two techniques that can be used to improve the low frequency room response of a reverberation room. One is to select room dimensions that will provide near uniform spacing of the room modes in the low frequency bands. The second technique is to install low frequency absorbers to the surfaces of the reverberation room to lower the "Q" of the individual modes. This paper will re-introduce a new quantity (first introduced at Internoise 2012 in New York City) called the modal distribution factor, D^2, and illustrate how it can be used to optimize the dimensions of any reverberation room within the space constraints of the proposed location. In addition, the paper will illustrate the design and acoustical performance of two very successful reverberation rooms that were conceived and constructed using these techniques.

9:00

3aAA4. Design of a new test chamber to measure the absorption, diffusion, and scattering coefficients. Peter D'Antonio (RPG Acoust. Systems LLC, 99 South St., Passaic, NJ 07055, pdantonio@rpgacoustic.com), Cheol Ho Jeong, and Mélanie Nolan (Danish Tech. Univ., Kgs. Lyngby, Denmark)

This presentation will describe the design for a new 384 m^3 test chamber that will allow the measurement of the random incidence absorption coefficient, according to ISO 354, the scattering coefficient, according to ISO 17497-1, the diffusion coefficient, according to ISO 17497-2 and the normal incidence low frequency absorption coefficient, according to ISO 10534-2. The rev room design incorporates optimal modal dimensional ratios in accordance with (T. Cox, P. D'Antonio & Avis, J. Audio Eng. Soc. Vol. 52, No. 6 (June 2004)), both boundary and hanging panel diffusers to allow satisfactory diffusivity, in collaboration with the research at DTU. Diffusivity will be verified using a reference absorber, consisting of 100 mm porous absorber with a 100 mm cavity. A sufficiently large reflection free volume is provided in the chamber for diffusion coefficient scale model measurements. The diffusion goniometer and 12 m^2 absorber samples are rotated in accordance with test. Scattering coefficient measurements are made according the ISO 17497-2 and the correlation method, using correlated polar responses. A steel and concrete lined low frequency impedance tube measuring 600 mm x 600 mm x 5.8 m long and 7 tons will be used to measure absorption from 25 to 250 Hz.

9:20

3aAA5. How to create an advanced acoustical test and research laboratory in 5 months. Ronald Sauro (NWAA Labs, Inc, 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com) and Jerry G. Lilly (JGL Acoust., Inc., Issaquah, WA)

This paper will take you through the need, design, building, and testing of a state of the art acoustical testing and research facility in a 5 month period. We will discuss the increasing need for wide frequency bandwidth measurements of absorption, transmission loss, very low level sound power measurements, and high resolution measurement of the directivity of speakers and diffusers. We will also look at the limits of standard laboratories and observe what changes are needed to extend those abilities and limits. At the end of the paper, we can look at the end result measurements of these facilities to see how close to design and performance the lab approached.

9:40

3aAA6. An investigation of laboratory sound transmission loss testing for steel-framed partitions. Benjamin Shafer (Tech. Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

Laboratory sound transmission loss (STL) test reports and research publications may be used to validate partition designs as valid and code-compliant sound isolation treatments in buildings. However, longitudinal and cross-laboratory analyses of the sound transmission loss of steel-framed building partitions have revealed such a wide variation in experimental results and calculated ratings that design validation is impossible. There are multiple possible explanations for this variation in results. Steel manufacturers list a variety of mil-thicknesses under the same "gauge" specification, for example. This research study investigates laboratory testing parameters and material components such as the repeated use of the same steel framing for varying panel configurations, screw length and spacing, and steel mil-thickness, regarding the effect that each of these aspects of sample assembly design and testing have on the sound transmission loss of steel-framed partitions.

10:00–10:15 Break

Contributed Paper

10:15

3aAA7. Acoustic characteristics comparison for glasses and windows by laboratory measurement. Hui Li, Jianghua Wang (Beijing Deshangjingjie Technol. Development Ltd. Co., Main Bldg., Rm. 104, Beijing, Beijing 100084, China, lihuisylvia@aliyun.com), Xiang Yan (Architecture, TsinghuaUniv., Beijing, China), and Jing Guo (Beijing Deshangjingjie Technol. Development Ltd. Co., Beijing, China)

Windows are usually with a much lower sound insulation index comparing to walls. Integrated sound insulation of building components is decided by the weakest element. So, windows are the key of indoor quietness. The Rw of 3 structures of glass and 5 structures of windows are measured in laboratory. By comparing the results, the influence of thickness of glass, thickness of air layer, thickness and material of lamination and frame are shown. Rw + Ctr of the glasses are between 34 dB and 39 dB. Rw + Ctr of the windows are between 27 dB and 41 dB.

Invited Paper

10:30

3aAA8. New ASTM ratings for quantifying low and high-frequency impact insulation in laboratory testing. John LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

The authors have recently published [*J. Acoust. Soc. Am.* **141**, pp. 428–-440 (2017)] a two-rating method for evaluating impact sound insulation, in which the low and high-frequency components of the floor are evaluated independently. Although originally described for field testing, the ratings can be applied to laboratory testing. ASTM draft classifications have been submitted for low and high-frequency versions of Impact Insulation Class, named LIIC and HIIC, respectively. The scope and use of the new ratings are explained.

Contributed Paper

10:50

3aAA9. Look—Do you see the noise leaking through that ceiling? Gary Madaras (Acoust., Rockfon, 4849 S. Austin Ave., Chicago, IL 60638, gary.madaras@rockfon.com)

Oftentimes, acousticians must convey complex, three-dimensional, acoustical phenomena that occur within and between rooms in buildings to architects, interior designers and other visually oriented people. The message can be lost during translation from quantitative acoustics metrics and their acronyms to the design and intersection of actual building elements such as walls and ceilings. The current phase of the Optimized Acoustics Research Program focuses on turning sound absorption and sound isolation performance visual by using color mapping. Much like a thermal imaging camera shows differences in surface temperatures, a sound intensity probe is being used to produce high definition color sound mapping of noise transmitting through acoustical ceiling systems and sound reflecting off or being absorbed by surfaces with different absorption coefficients. This measurement method helps to bridge the gap between the technical, quantitative side of acoustics and the visual side of design in an impactful and memorable way.

Invited Papers

11:05

3aAA10. Testing diffusion, Why we need to change the way we do things. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

For the last 20 years, different methods have been used in both educational and professional environments to provide test data regarding the performance of diffusers and diffusive products. In many cases, these test methods have defied certain recognized standards common in physics. In other cases, the test methods have been converted to use in test environments which are counter-intuitive to producing the outcomes expected. Additionally, the standard expectation of a single chart producing coefficient results in one or two lines has proven to provide far too little information for the acoustician. This minimal information also does not translate well to actual use in today's computer design programs. New test developments over the last 7 years by the author, in concert with others, have yielded test data that is both informational and multi-dimensional in nature. Providing both magnitude, polar, phase, and other data, this test, being developed under the guidance of ASTM, avoids the complications inherent in the current standards. This presentation will discuss both current standards and new technologies to overcome the limitations of those standards.

11:25

3aAA11. Using laboratory measurement data to improve acoustic simulations and evaluate performance. James J. DeGrandis (Acoust. First, 2247 Tomlyn St., Richmond, VA 23230-3334, jim@acousticsfirst.com), Hassan Azad (Univ. of Florida, Gainesville, FL), and Ronald Sauro (NWAA Labs, Inc., Elma, WA)

There are many different uses for the data that comes from a measurement. With the advent of new testing methodologies, data has become finely granular—allowing for precise analysis of the properties and performance of materials, geometries, and even simulations. Raw measurement data can be retained and compared directly to the output of complex acoustic models for development, improvement, evaluation, and eventually a form of calibration. Advances in technology are rapidly removing the limitations of computational power needed to create accurate models of acoustic phenomena in a timely and efficient manner. This allows a progression from limited particle simulations on single workstations, to large scale wave-based simulations using cloud-based GPU computing clusters. Performance will be evaluated, comparing their output to laboratory measurements, with the goal of creating better tools for acoustic design, prediction, experimentation, and development.

Contributed Paper

11:45

3aAA12. Density in diffusion, quantifying aural quality through testing and listening. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com) and Chips Davis (Chips Davis Designs, Concord, CA)

Diffusion in recording studios and critical listening spaces has been around for well over 30 years. During that time, acousticians and users alike confirm the fact that it makes a difference in the listening space but have a difficult time defining exactly what it is or why they like it. Diffusion test data in the past have been inconclusive as to showing what makes one design more pleasant to the listener than another. This is especially true with *in-situ* testing. The *in-situ* testing shown in the presentation will be concentrated on diffusion density or the amount and quality of diffuse energy being reflected. This presentation will also show a set of experiments between different sets of diffusers used as side diffusion in a recording studio. The experiments, using a binaural head and calibrated microphones, will show both test data as well as recorded responses which will be played back to the listening audience. Lab test data, using the proposed ASTM diffusion test under development, will be included to show the diffuser differences from a laboratory perspective. Attendees will be asked to contribute their perspectives to the recording playbacks to further the experiment as to the listener's experience.

Session 3aAB

Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Passive Acoustic Density Estimation: Recent Advances and Outcomes for Terrestrial and Marine Species I

Thomas F. Norris, Cochair
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Tiago A. Marques, Cochair
School of Mathematics and Statistics, University of St. Andrews, The Observatory, Buchannan Gardens, St. Andrews KY16 9 LZ, United Kingdom

Chair's Introduction—9:00

Invited Papers

9:05

3aAB1. Lessons learned from acoustically tracking baleen whales on the Navy's Pacific Missile Range Facility. Tyler A. Helble, E. E. Henderson (SSC-PAC, 2622 Lincoln Ave., San Diego, CA 92104, tyler.helble@gmail.com), Stephen W. Martin, Gabriela C. Alongi, Cameron R. Martin (National Marine Mammal Foundation, San Diego, CA), and Glenn Ierley (Univ. of California San Diego, Houghton, MI)

Detection, localization, classification, and tracking of marine mammals has been performed on the U.S. Navy's Pacific Missile Range Facility (PMRF) for over a decade. The range hydrophones are time-synchronized, have excellent spatial coverage, and monitor an area of approximately 1200 km^2. Even with these ideal assets, there are hidden challenges when attempting acoustic density estimates of baleen whales on the range. This talk will cover lessons learned from tracking several species of baleen whales on the range over the last decade.

9:30

3aAB2. Using detection received level as a guide for odontocete density estimation. John Hildebrand, Kaitlin E. Frasier, and Alba Solsona Berga (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu)

Odontocete echolocation clicks have been used for density estimation using cue-counting. We present an approach for assessment of the appropriate threshold for click detection based on the click received sound-pressure-level distribution. Under the assumption of random spatial distribution of echolocating animals with respect to the sensor, the number of detections should decrease with increasing received level. This pattern is verified by simulation modeling, even after incorporating diving behavior and sound production directionality. When measured received level distributions show peaks at values above the detection threshold, errors in the detection process are the most likely explanation. Potential remedies involve setting a higher detection threshold and/or fully characterizing both missed and false positive rates.

9:55

3aAB3. Parameters determining the suitability of bat species for acoustic monitoring. Jens C. Koblitz, Anne Scharf (Max Planck Inst. for Ornithology, Univ. of Constance, Constance 78464, Germany, jkoblitz@orn.mpg.de), Peter Stilz (BioAcoust. Network, Hechingen, Germany), Chloe Malinka (Aarhus Univ., Aarhus, Denmark), and Jamie D. Macaulay (Sea Mammal Res. Unit, Anstruther, Fife, United Kingdom)

Acoustic monitoring of bats is increasingly used in biodiversity assessments, population monitoring, and environmental impact assessments. In addition to accurate species identification, additional factors make it challenging to derive population trends or better sizes based on acoustic monitoring. Inter- and intra-species- as well as individual variation of acoustic parameters and acoustic activity result in varying detection probabilities. Changes in environmental conditions result in large changes in the volume monitored by the device. Differences in the devices used for acoustic monitoring make it inherently difficult to compare data collected with different devices. The single call monitoring volume is modelled for bats belonging to different guilds under consideration of the different call parameters such as call intensity, frequency, and directionality. By broadcasting bat echolocation calls from various distances to monitoring devices, the acoustic parameters influencing the successful detection of a call were examined. A microphone array was used to track bats in the vicinity of monitoring devices and the distance between device and bat was measured for each call based on the time of arrival difference. The acoustic detection function, the probability of detecting calls as a function of distance, was then derived for multiple detector types.

10:20–10:35 Break

10:35

3aAB4. Using passive acoustics to estimate populations of animals in groups. Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloepper@saintmarys.edu)

Passive acoustic monitoring is a widely used method to identify animal species and determine spatial and temporal activity patterns. One area where acoustic methods have not yet been successfully applied, however, is in determining population counts. Recently, my lab developed an acoustic method that accurately predicts the count of bats leaving from a cave roost in a given time period. Data were acquired of Brazilian free-tailed bats (*Tadarida brasiliensis*) across multiple nights and at different cave locations with different roost structures. We used a single microphone to monitor echolocation activity and simultaneously record the emerging bats with thermal and standard video. Bat abundance counts were determined from a single video frame analysis (every 10 sec) and were compared to different energy measures of an acoustic sequence recorded at the time of the analyzed video frame. For most cave locations, linear regression models successfully predicted bat emergence count based on acoustic intensity of the emerging stream, which indicates future population estimates may be collected solely with acoustics. Here, I describe this method and report on its application for counting other animals in groups.

10:50

3aAB5. Estimating population densities of temperate, insectivorous bats based on automatically recorded calls. Markus Milchram and Alexander Bruckner (Integrative Biology and Biodiversity Res., Univ. of Natural Resources and Life Sci., Gregor-Mendel-Strasse 33, Vienna 1180, Austria, markus.milchram@students.boku.ac.at)

Estimating population density based on automatically recorded calls is a key topic in bioacoustics, as individual recognition of animals is impossible. Several recently developed models do not require individual recognition but nevertheless allow to estimate density. However, there is a need to test these models on empirical data. Here, we used generalized random encounter models (gREM) to estimate population densities based on automatically recorded bat calls. To check the validity of the derived estimates, we fit Royle-Nichols models to species detection/non-detection data. Estimates of the two models were compared to each other and to estimates from published studies. The estimates of both models and literature estimates were within the same order of magnitude. Both models give reliable estimates of population density. However, we provide some cautionary notes for practical use: Bats which enter the detection sphere from above might bias results of gREMs, as the model simplifies the detection sphere to a two-dimensional area. On the other hand, reduction to detection/non-detection data in Royle-Nichols models results in information loss, which could limit their applicability in common species. Finally, we recommend to consider species behaviour carefully when applying one of the tested models.

11:05

3aAB6. Estimating density of fin whales and beaked whales with a mix distance sampling and spatially explicit capture recapture design. Tiago A. Marques, Len Thomas (School of Mathematics and Statistics, Univ. of St. Andrews, The Observatory, Buchannan Gardens, St. Andrews, Fife KY16 9 LZ, United Kingdom, tiago.marques@st-andrews.ac.uk), and Dave Moretti (NUWC, Newport, RI)

Dedicated passive acoustic density estimation studies with reliable survey design are still rare. We will present one such project: DECAF-TEA (Density Estimation for Cetaceans from Acoustic Fixed sensors in Testing and Evaluation Areas), which aims to estimate the density of beaked and fin whales on the SCORE US Navy range. We consider a mixed design that can deal with both species that produce acoustics signals with relatively short propagation distances (e.g., beaked whales) and those that produce signals with long propagation distances (e.g., fin whales), via distance sampling (DS) for the former and spatially explicit capture recapture (SECR) for the later. For beaked whales, we consider pairs of 3D bearing sensors to provide estimates of 3D localizations of detected whales, and from such data estimate detectability via distance sampling. At the same time, single units might provide a way to estimate density of fin whales via SECR. In either case, cue rates will be obtained from independent sources. In the context of DS we explore the impact of measurement error on the 3D bearing estimates and how this error propagates via errors in localization all the way through to potential bias in density estimation.

11:20

3aAB7. Song variability and change in the Australian pygmy blue whale song—Implications for passive acoustic abundance estimation. Capri D. Beck (Ctr. for Marine Sci. and Technol., Curtin Univ., Bentley, WA 6102, Australia, capridawn.jolliffe@postgrad.curtin.edu.au), Alexander Gavrilov (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, WA, Australia), Robert McCauley (Ctr. for Marine Sci. and Technol., Curtin Univ., Bentley, WA, Australia), Curt Jenner, and Micheline Jenner (Ctr. for Whale Res., Fremantle, WA, Australia)

Blue whales were heavily exploited throughout the early 20th century with many populations hunted to near extinction. Today, several known sub populations of blue whale exist, separated by geographic range and the acoustic signals they produce. The eastern Indian Ocean subpopulation of pygmy blue whales is easily identified by the production of a characteristic Australian song type. Long term collection of passive acoustic data from the Perth Canyon, Western Australia and Portland, Victoria, have revealed unprecedented levels of variability in the vocal behaviour of Australian pygmy blue whales. These levels of variability have significant implications for any attempts to assess the abundance of this population with acoustic techniques. Further, these findings may indicate higher levels of cognitive capability than previously expected in blue whales. The mechanisms behind variability in song production are unclear, though there is research to suggest that changes in vocal behaviour may be culturally driven or caused by changes in the ambient noise and environmental conditions. These findings highlight shortfalls in the current methods for measuring acoustic abundance, as well as indicate that higher than expected levels of variability may exist in the vocal behaviour of other pygmy blue whale subpopulations.

11:35

3aAB8. Learning deep features of spectrogram to estimate acoustic density of a fish species inhabitating *Posidonia oceanica* seagrasses. Lucas Lefèvre, Jean-Pierre Hermand, and Olivier Debeir (LISA - Environ. Hydro-Acoust. Lab, Université Libre de Bruxelles, av. F.D. Roosevelt 50, CP165/57, Brussels 1050, Belgium, lucas.lefevre@hotmail.com)

In field acoustic investigations on seagrass ecology, characteristic /kwa/ sound production of a fish species systematically present in meadows constituted by *Posidonia oceanica* species were recorded (Mediterranean, since 1995). Machine learning is investigated to automate estimation of acoustic density of the /kwa/ species. A multi-model deep neural network is implemented that directly inputs the spectrograms of signals recorded on hydrophones above and within the canopy. The network is a combination of 1D convolutional neural networks and a recurrent neural network with long short-term memory units and fully connected networks. Such architecture is able to process spectral information in the convolutional network and temporal information in the recurrent network, thereby making the best use of information in both time and frequency domains. In spite of high background noise level, the true-positive detection rate is shown to be excellent. With respect to human expert supervision, false-positive detections were, for the major part, faint sounds produced by distant fishes of the same species, as verified through careful re-listening. The apparent error rate is 8.4 $\times 10-^{2}$ on a partially mislabeled dataset of three days (7-10 pm). The true error rate, after label correction, is estimated at $1 \times 10-^{2}$.

3a WED. AM

Session 3aAO

Acoustical Oceanography, Underwater Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics: Arctic Acoustical Oceanography III

Peter F. Worcester, Cochair
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Mohsen Badiey, Cochair
University of Delaware, University of Delaware, Newark, DE 19716

Hanne Sagen, Cochair
Polar Acoustic and Oceanography Group, Nansen Environmental and Remote Sensing Center, Thormøhlensgt 47, Bergen 5006, Norway

Contributed Papers

8:00

3aAO1. Measurements of underwater noise north of Spitsbergen using deep-sea recorders. Dag Tollefsen and Helge Buen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no)

This paper presents results from year-long passive acoustic recordings in the seasonally ice-covered ocean north of Spitsbergen (Svalbard archipelago) from July 2016 to June 2017. Two moorings were deployed by FFI, each with an AMAR Ultra Deep acoustic recorder (JASCO Applied Sciences) equipped with a single hydrophone. The moorings were deployed and retrieved during open-water conditions and remained during periods of partially to near fully ice-covered conditions. We present results from analysis of the acoustic data for ambient noise spectra and statistics, and discuss characteristics of the spectra in relation to environmental factors including ice cover, wind and ocean waves. Seasonal noise spectra are compared with historic measurements from the eastern Arctic. Components of the sound field including transients due to ice, marine mammals, and anthropogenic noise will also be discussed.

8:15

3aAO2. Short-term fluctuations in the ambient noise field at an Arctic chokepoint horizontal line array. Dugald Thomson, David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., Rm. 3635 - 1355 Oxford St., Halifax, NS B3H 4R2, Canada, dugald@dal.ca), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and Garry J. Heard (Defence R&D Canada, Dartmouth, NS, Canada)

The Arctic is a region known for high variance in both ambient noise levels and local sound propagation conditions, and is currently experiencing an historic increase in shipping traffic, resulting in an acoustic environmental that is changing so rapidly that the sparse ambient noise data available from decades-old studies have been rendered obsolete. Renewed interest in Arctic acoustics have provided fresh noise data at northern sites, and in the summer of 2015 Defence R&D Canada collected a continuous two-week recording of the 48-hydrophone *Northern Watch* horizontal line array at Gascoyne Inlet. Over the ice-free study period, numerous ship passes, weather events, and anomalies were observed in the high-quality acoustic recordings. In this paper, spectral fluctuations, directionality, and correlation with ship tracks and weather observations in the noise data are analyzed to help improve the applicability of operational arctic noise models supporting sonar performance.

8:30

3aAO3. Modeling acoustic propagation in an ice-covered environment using a hybrid finite element-ray theory approach. Blake Simon and Marcia J. Isakson (Appl. Res. Labs. at the Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bsimon@arlut.utexas.edu)

Although models of acoustic propagation in an ice-covered environment have been calculated using only finite elements, ray theory models can cover longer distances and higher frequency ranges much more efficiently. In this study, a hybrid approach is taken by calculating the reflection coefficient of the ice using finite elements over a range of frequencies and angles. These values are then inserted into a ray-theory model. Using this model, a study of the sensitivity of common approximations of the ice cover on acoustic propagation is conducted. For example, the ice can be described simply as a pressure release surface or much more complexly as an elastic body with range and depth geo-acoustic property variations. The hybrid finite element-ray theory model will be verified numerically by comparing it to a full finite element propagation model for appropriate ranges and frequencies. [Work sponsored by ONR, Ocean Acoustics.]

8:45

3aAO4. Estimating channel capacity and sonar performance in the changing arctic. Paul C. Hines (Hines Ocean S&T Consulting, Dept. of Elec. and Comput. Eng., PO Box 15000, Halifax, NS B3H 4R2, Canada, phines50@gmail.com), Dainis Nams (GeoSpectrum Technologies Inc., Dartmouth, NS, Canada), Terry Deveau (JASCO Appl. Sci., Dartmouth, NS, Canada), Ron Kessel (Maritime Way Sci., Ottawa, ON, Canada), Jim Hamilton (JMH Consulting, Dartmouth, NS, Canada), Christopher Whitt (JASCO Appl. Sci., Dartmouth, NS, Canada), and David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Underwater acoustic processes including ambient noise, propagation, reverberation, and scattering have been studied for over half a century in the Canadian Arctic. Despite this, realistic predictions of communications and sonar performance have proved challenging due to the complex impact of ice cover on these processes, and the relative scarcity of sound speed profiles and information about the seabed. This challenge has been exacerbated in recent years because the rapid change in environmental conditions is reducing the relevance of many historical records. A key component of the present effort is to extract site specific model inputs from the environmental data contained in the literature, with an informed weighting toward more recent measurements. Site specific modelling was enhanced using inputs

from recent year-long ambient noise recordings. Model inputs were also influenced by specific source and receiver hardware being developed for use in future Arctic experiments. In this paper, low frequency sonar and communication performance estimates in the frequency band 20–250 Hz, based on site-specific environmental inputs and this low frequency hardware, are presented for several geographical regions of strategic relevance in the Canadian Arctic.

9:00

3aAO5. Low-frequency reverberation estimates using elastic parabolic equation solutions for ice-covered and ice-free Arctic environments. Scott D. Frank (Mathematics, Marist College, 3399 North Ave., Marist College Mathematics, Poughkeepsie, NY 12601, scott.frank@marist.edu) and Anatoliy Ivakin (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A computational model of reverberation at low frequencies in an ice-covered environment is developed. The model is built on a full-field perturbation approach and includes elastic parabolic equation solutions for the acoustic field and its horizontal and vertical derivatives near water-ice and water-air interfaces. Our previous work demonstrated that average reverberation intensity is sensitive to both elasticity and thickness of the ice at mid-frequencies, where the ice layer thickness is on the order of or larger than both compressional and shear wavelengths. Here we address what happens to the reverberation when ice thickness approaches zero. Reverberation estimates for rough free surface and those from rough ice-water interface with increasing ice layer thickness are compared to determine at what ice thicknesses and acoustic frequencies the long-range reverberation distinguishes between the two cases. To isolate effects of ice thickness, we assume roughness is the same in both environments. Frequencies will be varied from very low, where the ice layer is practically transparent, to moderately low where the presence of the ice becomes noticeable in the reverberation. Numerical examples for reverberation in a typical Arctic environment with upward refracting sound-speed profile are presented and discussed. [Work supported by ONR.]

9:15

3aAO6. A parallelization of the wavenumber integration acoustic modelling package OASES. Gaute Hope (Polar Acoust. and Oceanogr., Nansen Environ. and Remote Sensing Ctr., Thormøhlensgate 47, Bergen 5006, Norway, gaute.hope@gmail.com) and Henrik Schmidt (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA)

The wavenumber integration seismo-elastic model OASES can simulate the wave propagation in layered media, consisting of rough interfaces, elastic and porous layers. The wavenumber integration technique requires only the interesting frequencies, receiver depths, and offsets, to be calculated. As we seek to model pulse propagation at higher frequencies, denser sampling of frequencies must be applied. For complex media with many layers, and many vertical sections, the computation time quickly escalates. By exploiting that each frequency response can be calculated independently, a parallelization of the OASES package has been achieved and is presented here. This makes otherwise unfeasible or unpractical simulations computationally achievable. The parallel OASES package is applied to, and benchmarked on, several acoustic and seismic problems. The increased computation capacity is used to simulate and image the full wave field of several cases, reducing the computation time in one case from 1.5 years to 5 hours.

9:30

3aAO7. Inferring first-year sea ice thickness using broadband echosounders. Christopher Bassett (Resource Assessment and Conservation Eng., National Marine Fisheries Service, Alaska Fisheries Sci. Ctr., 7600 Sand Point Way NE, Seattle, WA 98115, chris.bassett@noaa.gov), Andone C. Lavery (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Jeremy P. Wilkinson (Br. Antarctic Survey, Cambridge, United Kingdom), Ted Maksym (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Zoe R. Courville (Cold Regions Res. and Eng. Lab., Hanover, NH)

Sea ice thickness is sparsely observed *in situ*. Under-ice acoustic techniques are a useful, but are generally used measure ice draft as a proxy for ice thickness. Using measurements of broadband acoustic backscattering (75–130 kHz) from laboratory-grown sea ice up to 80 cm thick, individual echoes from the water-ice and ice-air interfaces were isolated. Using the time delay between the echoes and a sound speed profile in the ice, total thicknesses were inferred and agree well with lengths of ice cores. This approach requires no ancillary data (i.e., water sound speed or pressure measurements). A temporal-domain model combining backscattering from both interfaces and from bubbles suspended within the brine channels also compares favorably with the measurements. This model is used to study the feasibility of applying the technique to first-year sea ice and to identify the bandwidths that best balance the constraints imposed by the scattering physics and practical considerations.

9:45

3aAO8. Measurements of sound speed in two types of ice with different microstructure. Anatoliy Ivakin, Wayne Kreider, and Bonnie Light (Appl. Phys. Laboratory, Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

Measurements of sound speed were made on laboratory-grown ice samples serving as proxies for two types of sea ice with different microstructure. The first one, congelation ice, has relatively homogeneous structure typical for first-year sea ice. The second one, granular ice, with a more heterogeneous microstructure, was prepared as a proxy for retextured multi-year sea ice. Both samples had approximately equal bulk salinity (12 ppt) to isolate effects of different microstructure. Sound speed was estimated by measuring time of flight of 400 kHz pulses and was found in congelation ice to be about 10% higher than in granular ice. Error bounds were estimated to be within 1% based on same measurements made for plastic material with known sound speed. Therefore, the results confirm sensitivity of sound speed in ice to its internal microstructure and heterogeneity which are known to affect large-scale ice properties, such as strength, elasticity, permeability, air-sea exchange, habitability, and partitioning of shortwave solar radiation. This motivates future work that would include modeling and measurement of sound speed, attenuation and scattering in natural sea ice, with a goal to develop a remote acoustic sensing technique for sea-ice characterization and, in particular, discrimination between first-year and multi-year ice types.

10:00

3aAO9. Underwater bubble low frequency source for Arctic acoustic tomography, navigation, and communication. Andrey K. Morozov and Douglas C. Webb (Teledyne, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

Arctic acoustic tomography requires a broadband source in the frequency range of 5–100 Hz. To meet this requirement, Teledyne Marine suggests applying a seismic source, recently developed for the Marine Vibrator Joint Industry Project. The source is considered to be less harmful for marine inhabitants and gives more precise and higher resolution imaging of the subsurface formations and structures. The coherent source is based on the application of an underwater, gas filled bubble resonator covered by an elastic membrane. The membrane supports high volume displacement. The sources are not sensitive to cavitation and to coupling effects. The fluid dynamics and acoustics of a spherical resonator are defined by the Rayleigh-Plesset equation. The buoyancy deforms the shape of a real bubble from spherical. The 3D simulation and experiments have shown that a cylindrical form is a practical engineering solution. It performances similar to a spherical bubble, keeps its shape and can be towed with a high speed. The Q-factor of a practical bubble resonator is ~10. Two systems are considered to cover a wide frequency band: a tunable resonator and a broadband multi-resonance system. The research proved that the bubble sound source is a practical solution for a frequency below 100 Hz.

Session 3aBAa

Biomedical Acoustics, Structural Acoustics and Vibration, and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications III

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Chair's Introduction—8:15

Invited Papers

8:20

3aBAa1. Bone guided-wave ultrasonography: How far are we from clinical implementation? Lawrence H. Le, Tho N. Tran, Kim-Cuong T. Nguyen (Dept. of Radiology and Diagnostic Imaging, Univ. of AB, 8440 - 112 St, Edmonton, AB T6G2B7, Canada, lawrence.le@ualberta.ca), and Mauricio D. Sacchi (Dept. of Phys., Univ. of AB, Edmonton, AB, Canada)

Bone guided-wave ultrasonography uses mechanical waves to study the long cortical bones. Long cortex has soft tissues above and marrow below acting like an ultrasound waveguide. The reverberations of the longitudinal and shear waves within the waveguide interfere constructively to generate energetic ultrasonic guided waves (UGW) travelling along the cortex. The UGW can be generated and recorded using an axial transmission technique with the transmitter and receiver deployed axially along the axis of the long bone on the skin's surface. The UGW thus acquired can be analyzed to provide information relevant to thickness and mechanical properties of the long bone. In this communication, we present an update of our research efforts on bone guided-wave ultrasonography including data acquisition, multichannel signal processing, bone modeling, and implementation of inversion algorithms to recover cortical thickness and mechanical parameters from UGW data. We also present some technical challenges, which will call for the joint effort of the bone ultrasound community to advance the science.

8:40

3aBAa2. High-resolution wavenumber estimation in ultrasonic guided waves using adaptive array signal processing for bone quality assessment. Shigeaki Okumura (Graduate School of Informatics, Kyoto Univ., Kyoto, Kyoto, Japan), Vu-Hieu Nguyen (Univ. of Paris-Est, Creteil, France), Hirofumi Taki (MaRI Co., Ltd., 22-44 Sakuragi-machi, Taihaku-ku, Sendai, Miyagi 982-0835, Japan, taki@marisleep.co.jp), Guillaume Haiat (CNRS, Creteil, France), Salah Naili (Univ. of Paris-Est, Créteil, France), and Toru Sato (Graduate School of Informatics, Kyoto Univ., Kyoto, Japan)

Quantitative ultrasound is a modality that is used to evaluate bone quality. It is considered that the analysis of ultrasound guided wave propagating along cortical bone may be useful for the assessment of cortical bone quality. Because the frequency-dependent wavenumbers reflect the elastic parameters of the medium, high-resolution estimation of the wavenumbers at each frequency is important. We report an adaptive array signal processing method with a technique to estimate the numbers of propagation modes at each frequency using information theoretic criteria and the diagonal loading technique. The proposed method estimates the optimal diagonal loading value required for guided wave estimation. We investigate the effectiveness of the proposed method via simple numerical simulations and experiments using a copper plate and a bone-mimicking plate, where the center frequency of the transmit wave was 1.0 MHz. An experimental study of 4 mm thick copper and bone-mimicking plates showed that the proposed method estimated the wavenumbers accurately with estimation errors of less than 4% and a calculation time of less than 0.5 s when using a laptop computer.

Contributed Papers

9:00

3aBAa3. Evaluation of axially transmitted shear waves in cortical bone: Experimental and simulation study. Leslie V. Bustamante Diaz, Masaya Saeki, and Mami Matsukawa (Dept. of Elec. and Electronics Eng., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0394, Japan, cyjb3302@mail4.doshisha.ac.jp)

Quantitative ultrasound has become a useful technique for bone assessment. Axial transmission technique has demonstrated various advantages over the through-transmission technique. Longitudinal wave and Lamb modes measurements in bone have been extensively reported while shear waves have been neglected. Shear wave velocities are lower than longitudinal wave velocities and can be useful for bone torsional strength characterization. In this study, plate samples were fabricated from bovine cortical bone. An ultrasonic pulse was transmitted along the long axis of the sample and shear waves were experimentally detected in the MHz range. Using time-of-flight technique, velocities were determined considering three different points of the received wave as a function of the ultrasound incident angle, transmitter-receiver distance and bone plate thickness. The observed shear wave velocities varied in a range (1630 to 1860 m/s) depending on the incident angle and the particular wave path, due to the anisotropy and heterogeneity of the bone structure. A 2D simulation using FDTD method was developed to understand the wave propagation phenomenon. Experimental and simulation results showed good agreement respect to the shear wave velocity estimated in a through-transmission measurement (1720 m/s), specially in angles close to the critical angle (60°).

9:15

3aBAa4. Model-based inversion of ultrasonic guided waves for cortical bone properties. Tho N. Tran (Dept. of Radiology and Diagnostic Imaging, Univ. of Alberta, Res. Transition Facility, Edmonton, AB T6G 2V2, Canada, tho.tran@ualberta.ca), Mauricio D. Sacchi (Dept. of Phys., Univ. of AB, Edmonton, AB, Canada), Dean Ta (Dept. of Electron. Eng., Fudan Univ., Shanghai, China), Vu-Hieu Nguyen (Laboratoire Modélisation et Simulation Multi Echelle UMR 8208 CNRS, Université Paris-Est, Creteil, France), and Lawrence H. Le (Dept. of Radiology and Diagnostic Imaging, Univ. of AB, Edmonton, AB, Canada)

Axial transmission ultrasonography, which uses a set of transmitting and receiving probes placed on the same waveguide's surface, shows the potential clinical application for cortical bone quality assessment. In this work, a model-based parameter sweep inversion approach has been developed to estimate the thickness and elastic velocities of the cortex from the dispersive axially-transmitted snapshots. The inversion algorithm is formulated in the frequency-phase velocity (*f-c*) domain. To solve the inverse problem, i.e., to extract bone properties from ultrasound data, a forward modeling has been developed to simulate the *f-c* dispersion curves given a bone model. A semi-analytical finite element (SAFE) method is used to compute the dispersion curves for a complex structure of a cortical bone plate coupled with overlying soft tissues. A parameter sweep is used to seek within a range of values an optimized solution with the least misfit. The proposed method optimizes the mismatch between the measured and theoretically calculated dispersion curves with a least-square constraint. Numerical and *in-vivo* experimental data examples are presented to illustrate the technique's performance.

9:30

3aBAa5. Inverse characterization of radius bone using low-frequency ultrasonic guided waves. Daniel Pereira (Mech. Eng., École de technologie supérieure, 100 Rue Notre-Dame O, Montreal, QC H3C 1K3, Canada, pereira.ufrgs@gmail.com), Julio Fernandes (Dept. of Surgery, Université de Montreal, Montreal, QC, Canada), and Pierre Belanger (Mech. Eng., École de technologie supérieure, Montreal, QC, Canada)

The success of the cortical bone characterization using the axial transmission is highly dependent on the inversion model used to match experimental data with the true bone properties. Simplified models such as plate or cylinder are typically used in the literature. In our previous work, a more elaborate model based on a bone-like geometry built using semi-analytical finite element (SAFE) method was introduced. The model was successfully implemented in an inverse characterization routine using laboratory-controlled measurements on bone phantoms. Thus, the aim of this work is to apply the proposed inverse scheme on the characterization of *ex-vivo* radius samples. In order to do so, five radiuses were taken from donors aged between 53 and 88 years old and tested using typical axial transmission configuration. Ultrasonic guided wave modes were excited at low-frequency (100-400 kHz) using a piezoelectric transducer and measured using a laser interferometer at the middle of the diaphysis. The measured velocities were systematically compared to velocities obtained with the SAFE model in order to predict the cortical bone properties. For each sample, four parameters were estimated: (1) Young's modulus; (2) density; (3) thickness; and (4) outer diameter. The results showed a notable correlation of the thickness and outer diameter with respect to the μCT images of the samples, while a less significant correlation was observed for the Young's modulus and density with respect to the gray level of the μCT images.

Invited Papers

9:45

3aBAa6. Effect of scattering and porosity on the apparent absorption in porous biphasic structures—Application to cortical bone. Yasamin Karbalaeisadegh, Omid Yousefian, and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

In this in-silico study, the attenuation coefficient in porous structures mimicking simplified cortical bone is calculated both in presence and absence of absorption, to isolate the effect of scattering on attenuation. Pore distributions with various pore concentrations (5–25 pore/mm^2) and diameters (40–120 μm) are generated. The propagation of 8 MHz plane waves is simulated using a 2D FDTD package. The attenuation coefficient is measured by fitting the exponential decay of the signal amplitude in the frequency domain during propagation in the absence or presence of absorption (coefficient of 10 dB/cm/MHz to mimic bone tissue), for different pore concentrations and sizes. The difference between total attenuation and scattering attenuation coefficients (obtained with no absorption) is defined as the apparent absorption coefficient. For ka<1 (k: wave number; a: scatterer diameter) the effect of absorption does not change with increasing porosity. However, for ka>1, increasing porosity leads to a higher apparent absorption, indicating a higher impact of absorption, even though the absorption coefficient and the overall volume fraction of absorbing material (solid phase) were kept constant. This suggests complex interactions between the wave and the structure, and could be attributed to longer propagation paths in the solid phase due to multiple scattering by the pores.

10:05–10:20 Break

Contributed Paper

10:20

3aBAa7. Characterizing porosity in bone-like porous media from frequency dependent attenuation. Rebekah White (Ctr. for Res. in Sci. Computation, North Carolina State Univ., Chapel Hill, NC), Omid Yousefian (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), H. T. Banks (Ctr. for Res. in Sci. Computation, North Carolina State Univ., Raleigh, NC), and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27695, mmuller2@ncsu.edu)

Osteoporosis affects both pore size and density in cortical bone. Quantifying levels of osteoporosis by inferring these micro-architectural properties from ultrasonic wave attenuation in cortical bone has yet to be done. Here, we propose a phenomenological power law model that captures the dynamics of frequency dependent attenuation in non-absorbing porous media mimicking a simplified cortical bone structure. We first demonstrate that, although it is frequency dependent, apparent absorption does not depend upon pore density and pore concentration, justifying the use of non-absorbing media for the simulations. We generate scattering attenuation data for various combinations of pore diameters (ranging from 20 to 100 μm) and pore densities (ranging from 3 to 10 pore/[mm]^2) using a 2D FDTD package (Simsonic), which simulates the propagation of elastic waves at frequencies of 1–8 MHz. The model is then optimized to fit these datasets by solving an inverse problem under an ordinary least squares (OLS) framework. With this we establish linear, functional relationships between the optimized power law model parameters and the micro-architectural parameters. These relationships showed that ranges of porosity could be inferred from attenuation data. Applying these techniques to attenuation data from cortical bone samples could allow one to characterize the micro-architectural properties of bone porosity.

Invited Papers

10:35

3aBAa8. Numerical simulation and machine learning based analysis of the ultrasonic waveform propagating in bone tissue. Yoshiki Nagatani (Dept. of Electronics, Kobe City College of Technol., 8-3, Gakuen-higashi-machi, Nishi-ku, Kobe, Hyogo 651-2194, Japan, nagatani@ultrasonics.jp), Shigeaki Okumura, and Shuqiong Wu (Graduate School of Informatics, Kyoto Univ., Kyoto, Kyoto, Japan)

Since bone has a complex structure, it is difficult to analytically understand the behavior of the ultrasound although it is useful for bone quality diagnosis. Our group, therefore, had been working on simulating ultrasound propagation inside bone models. In this paper, the results of the neural network based bone analysis using waveforms derived by the 3-D elastic FDTD (finite-difference time domain) simulation will be presented as well as its basis and some representative results of the FDTD method applied for the bone assessment. Since the FDTD simulation only requires the 3-D geometry of the model and the acoustic parameters (density, speed of longitudinal wave and shear wave) of the media, it has been very useful for evaluating each effect of a certain acoustic parameter or the bone geometry such as bone density, respectively, in addition to the purpose of visually understanding the wave behavior inside the model including the propagation path. Moreover, thanks to the recent powerful computational resource, it is now realized to prepare a huge number of waveforms for machine learning by using FDTD method. As a result, it was shown that the neural network can estimate the bone density better than the traditional method. (Grant: JSPS KAKENHI 16K01431.)

10:55

3aBAa9. Detection of fast and slow waves propagating through porous media. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg. 62, Rm. 2104, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov)

Porous media can support two longitudinal waves, which often overlap in time and frequency domains. Each wave has its own attenuation coefficient and phase velocity. These properties are related to volume fraction of solid phase, tortuosity, viscous characteristic length, and elasticity. Therefore, knowledge of individual wave properties is useful for characterizing porous media. Accordingly, methods for recovering separate reconstructions of fast and slow waves are of interest. The transfer function of a porous sample may be expressed as a weighted sum of two complex exponentials. The Modified Least Squares Prony's (MLSP) method may be used to recover the two individual components for non-dispersive media. Porous media are dispersive, however. The MLSP method may be augmented with curve-fitting (MSLP + CF) to account for dispersion. An alternative approach, based on Bayesian probability theory, is also powerful for recovering fast and slow waves. These approaches have been tested in through-transmission experiments on cancellous bone samples. Bayesian and MLSP + CF approaches successfully separated fast and slow waves and provided good estimates of the fast and slow wave properties even when the two wave modes overlapped in both time and frequency domains.

11:15

3aBAa10. Cancellous bone characterization using ultrasonic backscatter parametric imaging. Dean Ta, Ying Li (Dept. of Electron. Eng., Fudan Univ., 220 Handan Rd., Shanghai 200433, China, tda@fudan.edu.cn), Boyi Li (Dept. of Electron. Eng., Fudan Univ., Shanghai, Shanghai, China), Rui Zheng (School of Information Sci. and Technol., ShanghaiTech Univ., Shanghai, China), and Lawrence H. Le (Dept. of Radiology and Diagnostic Imaging, Univ. of Alberta, Edmonton, AB, Canada)

The ultrasonic imaging of the highly attenuated porous media (like spongy bone) is of great interest to the researchers. In this study, we used a 7.5 MHz focal transducer (V321, Panametrics, USA) to scan a total of 35 bovine cancellous bone samples. The −-6dB lateral resolution of the transducer was 0.157 mm which is comparable to the trabecular bone thickness (0.1–0.2mm). Apparent integrated backscatter (AIB) was calculated using different signal of interest (SOI) to reconstruct the ultrasonic backscatter parametric images. The trabecular bone structure can be clearly seen from the images. AIB was positively correlated with bone mineral density (BMD) ($R = 0.61$, $p<0.05$), bone volume fraction (BV/TV) ($R = 0.53$, $p<0.05$), and apparent bone density (ABD) ($R = 0.58$, $p<0.05$) when SOI was chosen around the surface of the bone sample, but negatively correlated with BMD ($R=-0.77$, $p<0.05$), BV/TV ($R=-0.81$, $p<0.05$), and ABD ($R=-0.82$, $p<0.05$) when SOI was chosen below the surface of the bone sample. We suppose that single scattering (SS) and multiple scattering (MS) may lead to this opposite correlation phenomena. When wavelength is comparable to the scatterer diameter, diffractive scattering is complex as predicted by Faran cylinder model. We suggest that MS may play an important role in cancellous bone characterization.

Contributed Paper

11:35

3aBAa11. Ultrasonic backscatter evaluates the variability of bone in pregnant women. Boyi Li, Dan Li, and Dean Ta (Dept. of Electron. Eng., Fudan Univ., Fudan University, Handan Rd. 220, Shanghai, 200433, China, Shanghai, Shanghai 200433, China, 16110720100@fudan.edu.cn)

Pregnancy and lactation-associated osteoporosis (PLO) is characterized by the deterioration of bone tissue and the fractures during pregnancy without any underlying disorders. This study attempted to investigate the variability of bone in pregnant women. The ultrasonic backscatter experiments were performed in Shanghai Changning Maternity & Infant Health Hospital. A total of 896 subjects were divided into four groups gestational age from 11th to 39th week, calculated apparent integrated backscatter coefficients (AIB) and spectral centroid shift (SCS). The estimated bone mineral density (Est. BMD) was measured using a medical ultrasonic transmission instrument. The body mass index (BMI) was calculated basing on height and weight. The results showed that AIB preformed a decreasing trend with the increasing of gestational age, the mean value in 37-39th-week group decreased 2.07 dB comparing with the 11-13th-week group, $p<0.001$. The SCS mean value in 37–39th-week group shifted 21.66 kHz than the 11–13th-week group, $p<0.001$. The Est. BMD mean value in 37–39th-week group decreased 0.05 g/cm^2 than the 11–13th-week group, $p<0.001$. The BMI increased with gestation age, the mean value in the 37–39th-week group was higher than the 11–13th-week about 4.19 kg/m^2, $p<0.001$. One of the reasons for these results was that pregnant women provide nutrition for the fetus, resulting in decreased bone tissue along with gestational age increasing. These results suggest that ultrasonic backscatter method might have a potential to diagnosis PLO.

WEDNESDAY MORNING, 7 NOVEMBER 2018 SIDNEY (VCC), 9:00 A.M. TO 11:50 A.M.

Session 3aBAb

Biomedical Acoustics and Physical Acoustics: Bubble Trouble in Therapeutic Ultrasound I

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Christy K. Holland, Cochair
Internal Medicine, Division of Cardiovascular Diseases and Biomedical Engineering Program, University of Cincinnati, Cardiovascular Center Room 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

3a WED. AM

Invited Paper

9:00

3aBAb1. Microbubble gas volume: A unifying dose parameter in blood-brain barrier disruption by focused ultrasound. Kang-Ho Song (Mech. Eng., Univ. of Colorado, 1111 Eng. Dr., campus box 427, Boulder, CO 80309), Brandon Harvey (Neurosci., NIH Bayview Res. Ctr., Baltimore, MD), and Mark Borden (Mech. Eng., Univ. of Colorado, Boulder, CO, mark.borden@colorado.edu)

Focused ultrasound with microbubbles is being developed to transiently, locally and noninvasively disrupt the blood-brain barrier (BBB) for improved pharmaceutical delivery. Prior work has demonstrated that, for a given concentration dose, microbubble size affects both the intravascular circulation persistence and extent of BBB opening. In this study, we independently measured the effects of microbubble size (2 vs. 6 μm diameter) and concentration, covering a range of overlapping gas volume doses (1–40 μL/kg). We first demonstrated precise targeting and a linear dose-response of Evans Blue dye extravasation to the rat striatum for a set of constant microbubble and ultrasound parameters. We found that dye extravasation increased linearly with gas volume dose, with data points from both microbubble sizes collapsing to a single line. A linear trend was observed for both the initial sonication and a second sonication on the contralateral side. Based on these results, we conclude that microbubble gas volume dose, not size, determines the extent of BBB opening by focused ultrasound (1 MHz, ~0.5 MPa at the focus). Finally, using optimal parameters determined for Evan blue, we demonstrated gene delivery and expression using a viral vector one week after BBB disruption.

9:20

3aBAb2. Persistent Pests: The role of gas diffusion in histotripsy bubble longevity. Kenneth B. Bader and Viktor Bollen (Dept. of Radiology, Univ. of Chicago, 5835 S. Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637, baderk@uchicago.edu)

Histotripsy is a focused ultrasound therapy under development for tissue ablation via the mechanical action of bubble clouds. While strong bubble activity is a hallmark of histotripsy, persistent bubble clouds shield the focal zone from subsequent pulses and prevent complete liquefaction of the target tissues. The diffusion of gas into the bubble during histotripsy exposure may be a contributing factor to the longevity of histotripsy-induced cavitation. To investigate the role of gas diffusion in bubble persistence, the oscillations of a single nanoscale nucleus exposed to a histotripsy pulse were computed in parallel with a first-order diffusion equation. The bubble gas content increased more than five orders of magnitude during the expansion phase, inflating the equilibrium bubble diameter by more than a factor of 50. The inertial collapse of the gas-filled bubble was arrested significantly in comparison to computations neglecting diffusion, as indicated by the minimum bubble size and maximum bubble wall speed during the collapse. The residual gas bubble required more than 15 ms for passive dissolution post excitation, consistent with experimental observation. These results indicate gas diffusion does not influence the bubble dynamics during the excitation, but is a contributing factor to bubble persistence between histotripsy pulses.

9:35

3aBAb3. Effect of high intensity focused ultrasound transducer F-number and waveform non-linearity on inertial cavitation activity. Christopher Bawiec (School of Medicine, Div. of Gastroenterology, Univ. of Washington, 325 9th Ave., Harborview Medical Ctr., Box 359634, Seattle, WA 98103, bawiec@uw.edu), Christopher Hunter, Wayne Kreider (CIMU, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), Vera A. Khokhlova, Oleg A. Sapozhnikov (CIMU, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Tatiana D. Khokhlova (School of Medicine, Div. of Gastroenterology, Univ. of Washington, Seattle, WA)

Enhanced chemotherapeutic drug delivery has been shown using pulsed high intensity focused ultrasound (pHIFU) without contrast agents. The threshold of these inertial cavitation effects has recently been correlated to the formation of shocks at the focus. The shock amplitude and corresponding peak negative pressure (p-) are primarily determined by the transducers F-number with less focused (higher F-number) transducers producing shocks at lower p-. Here, the dependence of inertial cavitation activity on F-number was investigated in gel phantoms using passive cavitation detection (PCD) and high-speed photography. HIFU transducers with the same aperture but different F-numbers (0.77, 1.02, and 1.52), operable at three driving frequencies (1 MHz, 1.5 MHz, and 1.9 MHz), were utilized with driving conditions consisting of 1 ms pulses delivered every second and p- from 1 to 15 MPa. Broadband noise emissions recorded by PCD were batch-processed to extract cavitation probability and persistence while concurrent imaging was performed in the focal pHIFU region. At the same p-, both PCD metrics and the imaging revealed enhanced cavitation activity at higher F-numbers. These results support the use of less focused, smaller-footprint transducers for achieving desired cavitation-aided drug delivery. [Work supported by NIH R01EB023910, K01DK104854, R01EB007643, and RFBR 17-54-33034.]

9:50

3aBAb4. Targeted microbubble opening of cell-cell junctions for vascular drug delivery elucidated with combined confocal microscopy and Brandaris 128 imaging. Ines Beekers, Merel Vegter, Kirby R. Lattwein, Frits Mastik, Robert Beurskens, Antonius F. W. van der Steen (Biomedical Eng., Erasmus MC, Office Ee2302, P.O. Box 2040, Rotterdam, Zuid Holland 3000 CA, Netherlands, d.beekers@erasmusmc.nl), Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Delft, Netherlands), Nico de Jong (Acoust. Wavefield Imaging, Delft Univ. of Technol., Rotterdam, Netherlands), and Klazina Kooiman (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands)

Ultrasound insonification of microbubbles enhances vascular drug delivery pathways, such as opening of cell-cell junctions and pore formation (sonoporation). However, the underlying mechanism remains unknown. The aim of our study was to elucidate the microbubble-cell interaction using the Brandaris 128 ultra-high speed camera (~17 Mfps), to visualize microbubble oscillation, coupled to a custom-build Nikon confocal microscope, to visualize cellular response. Confluent endothelial cells were evaluated for opening of cell-cell junctions with Cell Mask and for sonoporation with Propidium Iodide (PI). The cellular response of single targeted microbubbles ($n = 168$) was monitored up to 4 min after ultrasound insonification (2 MHz, 100–400kPa, 10-cycles). Cell-cell junctions opening occurred more often when cells were only partially attached to their neighbors (45%) than when fully attached (15%). Almost all fully attached cells showing cell-cell opening also showed PI uptake (92%). The mean microbubble excursion was larger when a cell was sonoporated (1.0μm) versus non-sonoporated (0.47μm). Additionally, larger microbubble excursion amplitudes correlated with larger pore size coefficients, obtained by fitting the PI uptake profiles to the Fan model [Fan, *et al.*, PNAS, 2012]. In conclusion, using the state-of-the-art imaging system we can now elucidate the relationship between microbubble oscillation behavior and the drug delivery pathways.

10:05–10:20 Break

10:20

3aBAb5. Prototypical madness: Design and demonstration of a magnetic-acoustic targeted drug delivery system. Michael Gray, Estelle Beguin, Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, IOxford OX37DQ, United Kingdom, michael.gray@eng.ox.ac.uk), Heather Nesbitt, Keiran Logan, Sukanta Kamila, Anthony McHale, John Callan (School of Pharmacy and Pharmaceutical Sci., Ulster Univ., Coleraine, United Kingdom), and Lester Barnsley (Heinz Maier-Leibnitz Zentrum, Jülich Ctr. for Neutron Sci., Julich, Germany)

Ultrasound, microbubbles, and magnetic nanoparticles have been used both separately and in varying combinations for targeted drug delivery. Recent studies have demonstrated the therapeutic benefit of magnetic microbubble (MMB) retention and acoustic targeting using separate devices. As a developmental step towards clinical implementation, a magnetic-acoustic device (MAD) was designed for the purpose of generating co-aligned magnetic and acoustic fields with a single hand-held enclosure. This paper presents *in vitro* characterization and *in vivo* demonstration of a targeted therapeutic system wherein the MAD non-invasively retains and activates drug-loaded MMBs. Free field experiments were conducted in order to characterize the magnetic field and its gradient for MMB capture, and to quantify acoustic field strength and directivity. Flow phantom experiments were used to quantify MMB retention and illustrate the resulting enhancement of cavitation activity. Murine experiments then demonstrated therapeutic efficacy in a pancreatic cancer model, showing a significant benefit in comparison to the use of separate magnetic and ultrasonic devices.

10:35

3aBAb6. Feeling gassy? Modifying the oxygen partial pressure of a fluid using acoustic droplet vaporization and different droplet concentrations. Haili Su, Karla P. Mercado-Shekhar, Rachel P. Benton (Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209), Bin Zhang (Div. of Biostatistics and Epidemiology, Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), Sneha Sharma, and Kevin J. Haworth (Univ. of Cincinnati, Cincinnati, OH, kevin.haworth@uc.edu)

Nucleating acoustic droplet vaporization reduces the dissolved gas content in a fluid. The objective of this study was to determine if the change in the partial pressure of oxygen (Po_2) could be modulated by adjusting the concentration of micron-sized perfluoropentane droplets. The droplets were manufactured using high-speed shaking and size-isolated using differential centrifugation (1 to 6 μm in diameter). Droplets were diluted in saline with a Po_2 of 154 mmHg and pumped through a 37°C flow phantom at 40 mL/min. The concentration ranged from 3.5×10^6 to 3.5×10^8 droplets/mL. A 5 MHz focused transducer insonified droplets at peak negative pressures of 4.05 MPa with a 500-Hz pulse repetition frequency and 5-cycle pulse duration. The Po_2 was measured downstream of the insonation region. At the lowest droplet concentration, the Po_2 was reduced to 129 mmHg. As the droplet concentration was increased, the Po_2 was reduced further. The reduction was in agreement (intra-class correlation and Pearson correlation coefficients greater than 0.9) with the model reported by Radhakrishnan *et al.* (2016, Ultrason. Sonochem.). At the highest droplet concentration, the Po_2 was reduced to 31 mmHg. These results demonstrate that ADV with varying droplet concentration modulates the oxygen partial pressure in a fluid.

10:50

3aBAb7. Phospholipid-coated microbubbles: Controlling response to ultrasound. Simone A. Langeveld (Biomedical Eng., Erasmus MC, Wytemaweg 80, Rotterdam 3000 CA, Netherlands, s.a.g.langeveld@erasmusmc.nl), Ines Beekers, Antonius F. W. van der Steen, Nico de Jong, and Klazina Kooiman (Biomedical Eng., Erasmus MC, Rotterdam, Zuid Holland, Netherlands)

Variability in response to ultrasound is an issue when microbubbles are used for drug delivery. This may be caused by immiscible phospholipid components and lipid phase separation in microbubble coatings. Since cholesterol influences phase distribution in lipid monolayers, we studied its effect (10 mol. %) on DSPC-based microbubbles. Lipid phase and ligand distribution of microbubble coatings were studied using super-resolution microscopy. Microbubble behavior upon ultrasound insonification (20 to 50 kPa, 2 MHz, single 10-cycle burst) was studied using the Brandaris 128 ultra-high-speed camera. As expected, lipid phase separation was observed in microbubble coatings without cholesterol. However, in microbubbles with cholesterol no lipid phase separation was observed. Without cholesterol, the ligand distribution was heterogeneous for only 3.9% of the microbubble surface (n = 42), while this was significantly more (11.6%) for microbubbles with cholesterol (n = 25), likely due to buckles. Moreover, cholesterol-containing microbubbles retained characteristic resonance behavior in response to ultrasound. These results demonstrate that although cholesterol in the microbubble coating affects the miscibility and ligand distribution of phospholipid components, a functional response to ultrasound is preserved. The phospholipid-cholesterol-coated microbubbles we developed may therefore have potential as theranostic agents. [Funding by the Phospholipid Research Center in Heidelberg, Germany, is gratefully acknowledged.]

11:05

3aBAb8. Multi-source passive acoustic source localisation. Catherine Paverd, Erasmia Lyka, and Constantin Coussios (Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom, catherine.paverd@eng.ox.ac.uk)

Accurate acoustic source localisation (ASL) has significant potential to improve diagnostic ultrasound imaging of small vessel structures. Recently, super-resolution imaging techniques for both active and passive source mapping have been developed; however, these approaches assume the presence of a single source within the point spread function (PSF) of the system. In reality, multiple sources may be present, for example, when high concentrations of contrast agent are used. We use a two-step approach to localise multiple sources within the PSF of a clinically relevant passive cavitation detection system. First, we apply a Blind Source Separation (BSS) technique known as Independent Component Analysis, which relies on higher-order statistics to separate and reconstruct signals originating from independent sources. Second, we determine the time-difference-of-arrival of the separated signals on each receiver, and perform ASL by fitting a polynomial using least squares regression. Simulation and experimental results demonstrate that with the BSS-ASL combination, multiple sources aligned axially within the PSF of a single array located 70 mm from the focus can be localised with sub-millimetre resolution. We verify sources as distinct using Passive Acoustic Maps generated from a perpendicular linear array. Further work is necessary to ascertain performance limits and to validate the technique *in vivo*.

11:20

3aBAb9. Non-linear acoustic emissions from therapeutically driven microbubbles. Paul Prentice and Jae Hee Song (CavLab, Medical and Industrial Ultrason., Univ. of Glasgow, Ninewells Hospital, Dundee DD1 9SY, United Kingdom, paul.prentice@glasgow.ac.uk)

Acoustic detection of contrast agent microbubbles, infused to the vasculature for exposure to focused ultrasound, is now routinely undertaken to evaluate therapy and avoid irreversible tissue damage. Harmonic, subharmonic (to the frequency of the focused ultrasound, f_0) and broadband emissions are often used to distinguish between stable and inertial cavitation activity, and associated bioeffects. The driven microbubble dynamic responsible for the generation of non-linear emissions, however, may not be well understood. Results from an investigation of single SonoVue microbubbles flowing through a capillary, for exposure to focused ultrasound at f_0 = 692 kHz, will be presented. Dual high-speed imaging from orthogonal perspectives at circa 2×10^5, and shadowgraphically at 10 million frames per second, capture microbubble activity and shock wave generation. Acoustic emissions are simultaneously collected with a calibrated broadband needle hydrophone, and high frequency imaging array on passive receive, for cavitation mapping. The results indicate that non-linear emissions are mediated by periodic bubble collapse shock waves, with subharmonic emission occurring above a threshold driving pressure amplitude. Implications for detection and quantification of driven microbubble cavitation activity, as well as conventional classification as stable or inertial, will be discussed. [This work was supported by ERC Starting Grant TheraCav, 336189.]

11:35

3aBAb10. Characterization of lipid-encapsulated microbubbles for delivery of nitric oxide. Himanshu Shekhar, Arunkumar Palaniappan (Dept. of Internal Medicine, Univ. of Cincinnati, 3933 Cardiovascular Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267, himanshu.shekhar@uc.edu), Cameron McDaniel, Daniel J. Hassett (Dept. of Molecular Genetics, Biochemistry & Microbiology, Univ. of Cincinnati, Cincinnati, OH), and Christy K. Holland (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Nitric oxide (NO) is a potent bioactive gas capable of inducing vasodilatory, anti-inflammatory, and bactericidal effects. The short half-life, high reactivity, and rapid diffusivity of NO make therapeutic delivery challenging. The goal of this work was to develop a technique to sequester NO within lipid-shelled microbubbles. Microbubbles loaded with either NO alone (NO-MB) or with NO and octafluoropropane (NO-OFP-MB) were synthesized by high-shear mixing of 1 mL lipids with either 1 mL of NO, or a mixture of 0.9 mL NO and 0.1 mL OFP. The size distribution and attenuation coefficient of NO-MB and NO-OFP-MB were measured using a Coulter counter and a broadband acoustic attenuation spectroscopy system, respectively. The payload of NO in the microbubbles was assessed using an amperometric micro-electrode sensor. Co-encapsulation of NO with OFP increased the number density, attenuation coefficient, and temporal stability of lipid-shelled microbubbles. However, the amount of NO loaded in NO-MB and NO-OFP-MB was similar (0.91 $\pm$ 0.03 μM and 0.93 $\pm$ 0.1 μM, respectively). These results suggest that NO can be encapsulated within lipid-shelled microbubbles. However, co-encapsulation of OFP does not enhance the NO payload or the temporal stability, despite an increase in attenuation and number density of lipid-shelled microbubbles.

3a WED. AM

Session 3aED

Education in Acoustics: Hands-On Demonstrations

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

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 Victoria. 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 effor 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)

WEDNESDAY MORNING, 7 NOVEMBER 2018 CRYSTAL BALLROOM (FE), 9:45 A.M. TO 12:00 NOON

Session 3aMU

Musical Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Percussion Instruments

Uwe J. Hansen, Cochair
Indiana State University, 64 Heritage Dr., Terre Haute, IN 47803-2374

Andrew C. Morrison, Cochair
Natural Science Department, Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Invited Papers

9:45

3aMU1. The state of percussion research in musical acoustics: How we got here and where it is going. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

Drums are among the most ancient of all musical instruments and have been found in nearly all cultures across the world. This work covers a selection of major investigations of the acoustics of percussion instruments by a variety of scientists throughout history. Ernst Chladni's efforts to characterize the vibration of plates are well known in acoustics and often replicated by students in the laboratory. Other scientists who made early contributions to the understanding of percussion instruments include Lord Rayleigh and C. V. Raman. Rayleigh's observations of the acoustics of the kettledrum and bells represented only a fraction of his contributions to acoustics. Raman, who is remembered more for his work in spectroscopy, made detailed studies of the traditional Indian drum, the tabla. Modal analysis techniques used to investigate the acoustics of percussion instrument have evolved significantly in recent years. A brief review of current modal analysis techniques for studying percussion instruments is presented in this work. In addition, suggestions for future directions of percussion research is described here.

10:05

3aMU2. Musical drumhead damping using externally applied commercial products. Randy Worland and William Miyahira (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Snare drums and tom toms used in drum sets often produce an undesirable high frequency ringing sound when struck. Drummers have historically applied DIY damping solutions to address this ringing, placing objects such as wallets and tape on the heads. A large variety of inexpensive commercial products is now available to dampen drums in a more controlled manner. The most commonly used products consist of small adhesive pads that can be placed directly on the drumhead at desired locations, either singly or in combinations. An experimental investigation of the ringing drumhead modes and the effects of the damping pads is reported. Decay rate

measurements indicate that the (3,1), (4,1), and (5,1) modes often contribute the most to the ringing sound, and that these modes are strongly affected by the damping pads. Another class of commercial dampener consists of thin annular rings of Mylar that conform to the perimeter of the drum head. These free-floating rings produce a similar damping effect, but dissipate energy through a different physical mechanism. Experimental results for both types of dampeners are presented and discussed in terms of viscoelastic and air-layer damping mechanisms, respectively.

10:25

3aMU3. Steel pan developments. Uwe J. Hansen (Phys., Utah Valley Univ., 6104 N Lake Mt Rd., Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu) and Thomas Rossing (CCRMA, Stanford Univ., Los Altos Hills, CA)

The Caribbean steel pan is likely the single most significant new acoustic musical instrument of the 20th Century. Some major developments incorporated by Felix Rohner of Panart have led to important instrument modifications. Among the differences to be discussed are as follows: using specified steel alloys in the sheet metal to replace commercial 55 Gal drums, sinking the pan in a press rather than by hand, surface hardening the playing surface in a nitride bath, dispensing with chiseled note section boundaries, adding a central dome to each note section, replacing the pan by a Hang (a lap-held instrument played by hand), and finally an additional air volume enclosed, to enhance low frequency resonances (the Gubal).

10:45–11:05 Break

11:05

3aMU4. Take me out to the ballgame: The acoustics of the softball bat piano. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dar119@psu.edu)

As a multimedia supplement for a recent article "Acoustics and Vibration of Baseball and Softball Bats" [*Acoustics Today*, **13**(4), 35–42 (2017)] the author created a "softball bat piano" and played the tune "Take Me Out to the Ballgame." The bat piano is a collection of softball bats, clamped at the handles, selected for their barrel frequencies. It is played by striking the bat barrels with a softball. The structural modes of the hollow cylindrical barrel of an aluminum or composite bat are responsible for the trampoline effect which affects the performance of the bat as it is used for the game of softball. However, the "hoop mode" which gives rise to the trampoline effect is also the source of the bat piano's audible musical sound. The nature of these cylindrical shell vibrations will be discussed, highlighting similarities to the musical properties of bells and chimes. The influence of bat construction and materials on the "hoop frequency" will be explained, along with the process of selecting bats with a range of frequencies wide enough to form a musical scale. If time permits, video of additional musical selections will be shown.

11:25

3aMU5. How beatboxers produce percussion sounds: A real-time magnetic resonance imaging investigation. Timothy Greer (Signal Anal. and Interpretation Lab., Univ. of Southern California, Los Angeles, CA 90007, timothdg@usc.edu), Reed Blaylock (Dept. of Linguist, Univ. of Southern California, Los Angeles, CA), Nimisha Patil, and Shrikanth S. Narayanan (Signal Anal. and Interpretation Lab., Univ. of Southern California, Los Angeles, CA)

Beatboxing is a musical artform in which performers use their vocal tract to create percussion sounds. Sometimes beatboxers perform as a part of an ensemble, using their vocal tract to provide a beat for other musicians; other times, beatboxers perform alone, where they might sing and produce percussion sounds simultaneously. We present methods in real-time magnetic resonance imaging (rtMRI) that offer new ways to study the production of beatboxing sounds. Using these tools, we show how beatboxers can concatenate intricate articulations to create music that mimics the sound of percussion instruments and other sound effects. The rtMRI methodology reveals how different beatboxers play their vocal folds to perform characteristic "clean" or breathy styles. By using rtMRI to characterize different beatboxing styles, we show how video signal processing can demystify the mechanics of artistic style.

Contributed Paper

11:45

3aMU6. Preliminary modal analysis of the Alo Gong. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com), Tracianne B. Neilsen (Brigham Young Univ., Provo, UT), and Maria T. Keke (Music, Univ. of Nigeria, Nsukka, Nigeria)

The Alo is a percussion instrument found predominantly in the eastern parts of Nigeria with the Igbo's and plays an important role in the Igbo musical culture and settings. Being a bass gong with tones of three pitches, the Alo produces a very distinct sound and a sustained reverberation rendering a background beat to music of an African instrument ensemble or a mixed ensemble. This paper discusses the acoustical output characteristics of this instrument. The spectral characteristics of the sustained reverberation is presented, along with the dynamics responses of the structure to an excitation. These analyses not only provide an understanding of the distinct sound of the Alo instrument but may also be used by African instrument acoustician to re-model, or create more instruments with resonance frequencies.

Session 3aNS

Noise, ASA Committee on Standards, and Signal Processing in Acoustics: Technological Challenges in Noise Monitoring

Matthew G. Blevins, Cochair
U.S. Army Engineer Research and Development Center, 2902 Newmark Drive, Champaign, IL 61822

Anton Netchaev, Cochair
U.S. Army Engineer Research and Development Center, Vicksburg, MS

Chair's Introduction—8:00

Invited Papers

8:05

3aNS1. Benefits and challenges of using consumer audio equipment for unattended acoustical monitoring. Daniel J. Mennitt (Elec. and Comput. Eng., Colorado State Univ., 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, daniel_mennitt@partner.nps.gov), Damon Joyce, and Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

The quality of an acoustical measurement impacts the accuracy of all inferences that rely on the resulting data. While standard sound level meters are well suited for noise studies requiring high precision, their cost, power consumption, and capabilities constrain the scope of application. Alternatively, the wide variety of consumer audio equipment offers many options for acoustical monitoring. The ability to make high resolution, multichannel audio recordings with packages that are relatively small, inexpensive, and low power is especially attractive for long-term acoustical monitoring in remote areas and large-scale spatial surveys that require many devices. These recordings are more valuable when they are calibrated and processed to yield sound level data. Despite the promise of consumer audio equipment, there are several drawbacks and unknowns. Within the framework of the signal chain, this talk will discuss some of the benefits, challenges, and unknowns of using consumer audio equipment for unattended acoustical monitoring.

8:25

3aNS2. Weather focused challenges for continuous monitoring of military noise. Jordan D. Klein, Steven Bunkley, Sahil G. Patel, Richard D. Brown, Jason D. Ray (U.S. Army Engineer Res. and Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, jordan.d.klein@usace.army.mil), Jesse M. Barr, Matthew G. Blevins, Gregory W. Lyons (U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and Anton Netchaev (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS)

Domestic military installations generate high levels of noise due to testing and training which leads to annoyance and complaints from surrounding communities. This necessitates continuous noise monitoring to provide decision makers with the information they need to proactively manage their noise environment. Due to the diverse climates in which military testing and training are conducted (e.g., desert, tundra, and rainforest), monitoring equipment that can operate in a variety of environmental conditions with minimal maintenance and low power consumption is needed. Using existing technologies as a baseline, various iterations of a low-cost acoustic monitor were designed to meet these constraints while minimizing initial investment cost, improving the mean time between failures, and increasing overall system capability. This paper will describe the system developed to provide a rapid deployment option that is robust to extreme temperatures, humidity, and destructive wildlife. A review of operational logs collected during multiple deployments was used to evaluate system performance against benchtop and off-the-shelf solutions. This data demonstrate the reliability of the monitoring stations and the sustainability of their hardware.

8:45

3aNS3. Developments in continuous unattended monitoring systems. Douglas Manvell, Ken Anderson (EMS Bruel & Kjaer, Naerum, Denmark), and Bryce Docker (EMS Bruel & Kjaer, 2330 East Bidwell St., Ste. 210, Folsom, CA 95630, bryce.docker@emsbk.com)

Continuous, unattended noise monitoring systems can immediately alert you should noise levels exceed defined criteria. Once alerted to an exceedance, operators can act to return levels to compliance. This approach has two significant limitations. First, the operator can only take action after the breach has occurred and therefore systems are only able to inform owners about problems that have occurred in the past, rather than allowing them to maintain compliance. Second, the noise limit exceedances might not be due to specific noise from the operator but from unrelated, residual noise in the often-complex noise climates around the particular site and will then be the cause for a false positive. Compliance breaches are frequently triggered by aircraft overflights, road traffic or community sources. Modern monitoring systems enable users to view noise characteristics and listen to the noise breach to determine the source and act if

necessary. However, this approach can create a significant number of false positives each taking up operator time to address. A previous paper by the authors described how airport noise management systems have addressed this problem by combining data from other systems, and how different techniques are required in urban & industrial noise management. This paper describes developments in these techniques and gives examples of techniques that allow operators to take action before a compliance breach occurs, and to reduce the number of false positive alerts.

9:05

3aNS4. Rotorcraft and unmanned aerial system noise measurement technology development and challenges. James H. Stephenson (AMRDC US Army, 2 North Dryden St., MS461, Hampton, VA 23681-2199, james.h.stephenson@nasa.gov), Keith Scudder (AMA, Inc., Hampton, VA), and Eric Greenwood (NASA LaRC, Hampton, VA)

Rotorcraft and Unmanned Aerial Systems (UAS) produce undesired noise that propagates into the community. Significant research has been dedicated over the years to accurately predict the acoustic field of these vehicles from first principles. Despite significant advances in predictive capability, (semi)empirical predictions incorporating acoustic flight test measurements remain the most accurate way to model their noise. However, flight test measurements present significant challenges that must be mitigated. Examples include ground and atmospheric attenuation, inclement weather, and the need to ensure a low background noise condition at the test location. Background noise requirements typically limit testing to remote areas where facility power is not natively available. In order to help mitigate these challenges, a radio-controlled and ground-based measurement system with low space, weight, and power requirements was developed. This system, called Wireless Acoustic Measurement System II (WAMS II), is remotely activated, can record local weather and acoustic measurements with 24-bit accuracy, and has an extended battery life. These capabilities are an improvement on the first generation system and allow for the easy deployment of the multiple systems required to capture the full vehicle acoustic directivity patterns. The design, development, and effectiveness of these systems will be discussed.

9:25

3aNS5. A broadband impulsive signal detection filter for unknown acoustic sources in outdoor environments using non-coplanar microphone arrays. Steven Bunkley (Engineer Res. and Development Ctr., US Army Corps of Engineers, 3909 Halls Ferry Rd., Vicksburg, MS 39180, steven.l.bunkley@usace.army.mil), Michael J. White, Matthew G. Blevins (Engineer Res. and Development Ctr., US Army Corps of Engineers, Champaign, IL), Anton Netchaev, Jordan D. Klein (Engineer Res. and Development Ctr., US Army Corps of Engineers, Vicksburg, MS), Gregory W. Lyons (Engineer Res. and Development Ctr., US Army Corps of Engineers, Champaign, IL), and richard haskins (Engineer Res. and Development Ctr., US Army Corps of Engineers, Vicksburg, MS)

This paper describes a method for non-coplanar microphone arrays that temporally isolates and cleans unknown broadband acoustic impulses for detection, classification, and scene analysis. Possible events are initially identified using a sliding statistical time window. Then the authors posit that most of the false triggers due to environmental noise can be filtered by using generalized cross correlation to phase align the microphone channels and reject implausible velocities. Finally, the phase aligned signals are calibrated and averaged across the microphones. With appropriate hyperparameter tuning, this method appears robust to ambient noise, wind noise and physical interaction. Performance is measured using a simulation and a real historic dataset of over 2 hours of curated acoustic recordings containing 559 gunshots, 120 blasts, and 747 other various weather and non-impulsive events recorded with no prior information under normal operating conditions. Events were found and validated using human listeners with a tool to visualize the waveform and the spectrogram. For this dataset, the model accurately found over 95% of the gunshots with 92% temporal separation and 100% of the blasts identified by the listeners. These results show the method to be a viable solution for impulsive outdoor broadband acoustic signal detection.

Contributed Papers

9:45

3aNS6. Supervised machine learning for crowd noise classification at collegiate basketball games. Kolby T. Nottingham, Katrina Pedersen, Xin Zhao, Brooks A. Butler, Spencer Wadsworth, Blake Smith, Mark K. Transtrum, Kent L. Gee, and Sean Warnick (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT, katrina.pedersen@gmail.com)

Acoustical monitoring combined with machine learning (ML) may help in understanding crowd dynamics. While ML has been applied in numerous audio applications, the aim is usually to distinguish events from noise, rather than trying to characterize the noise itself. This paper comprises an initial study using ML to characterize crowd dynamics during collegiate basketball games. High-fidelity crowd noise recordings from several men's and women's games were synchronized with game video and used to produce a training dataset for supervised ML by linking game events (e.g., baskets, fouls) with acoustic labels (e.g., cheering, silence, and applause). Using the training dataset, a ML classifier was built to identify causal game events from acoustic crowd responses. Findings, potential improvements, and additional crowd noise applications are discussed.

10:00–10:15 Break

10:15

3aNS7. Exploring acoustical approach for pre-screening of building envelope airtightness level. Umberto Berardi (DAS, Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca)

Air infiltration plays a significant role in designing and evaluating the performance and air quality of a building. Air leakage through an existing building enclosure can both be localized and calculated by using experimental measurement, such as blower door test system, tracer gas method, and transient approach. Estimating building air permeability through these methods can be expensive, time consuming, and weather reliant. The economical and environmental effect of air infiltration through building envelope leads towards more research on detecting air leakage locations and estimating air infiltration rate through newly introduced techniques, such as using an acoustical method. As such, in this research, a general review of airtightness detection and quantification method will be presented, especially acoustical technique which will be explored more deeply. Moreover, due to significant impact of window systems on total air infiltration through building envelope, the correlation between sound transmission loss and air permeability through seven window assemblies in an existing building will be explored in order to investigate acoustical method further. In addition, the acoustic air leakage detection method based on standard ASTM E1186 will be

instigated on the experimental windows. The results reveal the negative relationship between these two phenomena.

10:30

3aNS8. Counting and tracking vehicles using acoustic vector sensors. Józef Kotus (Gdansk Univ. of Technol., Gdansk, Pomorskie, Poland) and Andrzej Czyzewski (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl)

A method is presented for counting vehicles and for determining their movement direction by means of acoustic vector sensor application. The assumptions of the method employing spatial distribution of sound intensity determined with the help of an integrated 3D intensity probe are discussed. The intensity probe developed by the authors was used for the experiments. The mode of operation of the algorithm is presented in conjunction with noise characteristics produced by moving vehicles. The optimization of the algorithm is based on measurements of intensity of sound emitted by the vehicle under controlled conditions. A test setup was built for this purpose with the use of measuring devices installed along a road with varying traffic flow. Reference data on the number of vehicles and traffic directions were prepared employing a recorded video and a reference traffic analyzer operating in lidar technology. It is shown that the developed acoustic method may contribute to an increase of effectiveness of commonly used vehicle counting systems employing inductive loops or Doppler radars. [Project financed by the by the Polish National Centre for Research and Development (NCBR) from the European Regional Development Fund under the Operational Programme Innovative Economy No. POIR.04.01.04-00-0089/16 "INZNAK—Intelligent road signs...".]

10:45

3aNS9. Mitigating the pitfalls of testing unmanned aerial system vehicles and components in anechoic chambers. James H. Stephenson (AMRDC US Army, Hampton, VA), Daniel Weitsman (Univ. of Hartford, MS461, Hampton, VA 23681, WEITSMAN@hartford.edu), and Nikolas S. Zawodny (NASA LaRC, Hampton, VA)

Flow recirculation is known to develop inside a closed anechoic chamber when testing unmanned aerial system (UAS) rotor components and vehicles. This flow recirculation modifies the inflow through the vehicle's rotors, which results in significant impacts to the measured acoustic signature. A measurement campaign was undertaken at NASA Langley Research Center in which a UAS rotor was tested inside a small anechoic wind tunnel. Acoustic signatures were obtained with the downwash exhausting down the wind tunnel, and with the tunnel exhaust plugged forcing flow recirculation. Several methods to mitigate the acoustic impacts of flow recirculation were then employed with the wind tunnel in the plugged configuration. The effectiveness of the mitigation strategies are discussed, along with implications for future testing and standards development.

11:00

3aNS10. Military and police small arms firing range noise assessments. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

Firing ranges are by their nature sources of very loud noise that can damage the hearing of on-site shooters and observers while also potentially causing a noise disturbance off-site. Small arms firing ranges are typically indoor ranges when located in urban or suburban settings, but when located in rural settings, they are typically outdoor ranges designed with varying types and degrees of noise abatement or no noise abatement at all, depending on the setting. Public firing ranges can be busy and in continuous use under good lighting and favorable weather conditions. Often there are wait times. Military and police (M&P) firing ranges are typically active under scheduled usage times. Both M&P ranges host a variety of small arms, from fully automatic and semi-automatic weapons to bolt-action sniper rifles and large caliber side arms, all of which are very loud. It is often mandatory for M&P personnel to undergo shooting practice on a regular basis with some assigned personnel having certification or qualification performance requirements. Several examples of military and police firing ranges, with their issues, similarities and differences, are discussed. Personnel, interior and/or exterior noise abatement measures are usually necessary for these indoor and outdoor M&P firing ranges.

11:15

3aNS11. Reactive acoustic liner design. Ramani Ramakrishnan (Architectural Sci., Ryerson Univ., 325 Victoria St., Toronto, ON M5B 2K3, Canada, rramakri@ryerson.ca) and David H. Van Every (Aiolos Eng., Toronto, ON, Canada)

Acoustic treatment to reduce fan noise levels in a wind tunnel circuit consists of fibrous materials such as fibreglass or rockwool. Open cell foam materials are also used as acoustic treatments. The acoustic treatments are conventionally applied at fan tail cone regions, tunnel walls along fan diffuser section, cross-legs, test section diffuser, and nozzle contraction areas. However, conventional treatments are not possible in cryogenic wind tunnels, since bulk absorber materials with required resistivity, when operating at cryogenic temperatures, are not available. One possible solution is to design reactive silencers tuned to dominant frequencies. One such approach was used as noise control technique so as to satisfy test section noise specifications. The sound power spectrum of the compressor at different speeds were evaluated. The estimated test section sound pressure levels showed noise reduction at two dominant frequencies were required. The acoustic treatment, therefore, resulted in a double layer reactive design tuned to the two dominant frequencies. The design process will be highlighted in the presentation. The final treatment details will also be presented.

WEDNESDAY MORNING, 7 NOVEMBER 2018 RATTENBURY A/B (FE), 8:30 A.M. TO 11:45 A.M.

Session 3aPA

Physical Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics: Willis Coupling in Acoustic Metamaterials

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Feruza Amirkulova, Cochair
Mechanical Engineering, Western New England University, 1215 Wilbraham Road, Springfield, MA 01119

Invited Papers

8:30

3aPA1. Bianisotropic acoustic metasurfaces for wavefront transformation. Steven Cummer, Junfei Li, Chen Shen (Dept. of Elec. and Comput. Eng., Duke Univ., PO Box 90291, Durham, NC 27708, cummer@duke.edu), Ana Díaz-Rubio, and Sergei Tretyakov (Dept. of Electronics and NanoEng., Aalto Univ., Aalto, Finland)

Acoustic metasurfaces are thin, engineered structures that can control the local reflection and transmission phase of acoustic waves, and thus enable a high degree of flexibility for wave manipulation. Here, we describe our recent work on so-called perfect metasurfaces, in which elements are designed that can control the local transmission and reflection amplitude and phase response of the metasurface. Controlling both the amplitude and the resulting asymmetric phase response requires a fundamentally asymmetric surface impedance for the unit cells. The resulting ideal surface properties thus have strong links to the concept of bianisotropy in electromagnetics and Willis coupling in elastodynamics. We have developed a shunt resonator-based element design that contains the needed degrees of freedom to independently control the surface impedance on both sides of the metasurface. Using this perfect metasurface design approach, we demonstrate the design and experimentally measurement of high efficiency wide-angle transmissive beam steering at levels beyond what is possible with phase control alone.

8:50

3aPA2. Willis coupling in underwater elastic Helmholtz resonators. Xiaoshi Su and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, xiaoshi.su@rutgers.edu)

Helmholtz resonators (HRs) have been central in many acoustic metamaterial devices. For example, one can use an array of HRs to obtain negative effective bulk modulus or a panel of HRs to achieve total absorption of low-frequency sound. The aforementioned applications are based on the local monopolar resonance of each HR. However, the HR behaves differently in water. The elastic modulus of the resonator wall can become comparable to the modulus of water making the resonant frequency much lower than the rigid wall case. Moreover, considering the wall elasticity and mass leads to large structural asymmetry and induces cross-coupling between pressure and velocity fields. In this talk, we provide a lumped element model in order to predict the resonant frequency of the elastic HR. The model is demonstrated by sound scattering from an elastic HR in the context of acoustic bianisotropy, or Willis behavior, following a recent paper [Li Quan *et al*., Phys. Rev. Lett., 2018]. It is found that both the pressure and velocity fields can generate monopole and dipole responses from an elastic HR. The explicit example of an elastic HR in a one-dimensional waveguide will be discussed.

9:10

3aPA3. Dynamic homogenization of spatio-temporal metamaterials. Daniel Torrent (Phys., Universitat Jaume I, Av. de Vicent Sos Baynat, s/n, Castellon de la Plana, Castellón 12071, Spain, dtorrent@uji.es)

We discuss the homogenization of acoustic metamaterials with spatio-temporal modulation. It is shown that the effective medium required for the correct description of these structures is a non-recyprocal Willis material. Analytical expressions are given for the effective parameters, and several numerical examples are shown. The theory is applied as well to the specific case of thermal metamaterials, and it is shown that a thermal diode is feasible by spatio-temporal modulation.

9:30

3aPA4. Wave propagation in 1D piezoelectric crystal coupled with 2D infinite network of capacitors. Anton A. Kutsenko (Appl. Mathematics, Jacobs Univ., Campus Ring 1, Bremen 28759, Germany, a.kutsenko@jacobs-university.de)

For 1D piezoelectric crystal, the dispersion equation is a simple linear dependence between the frequency and the wave number. After connecting the crystal to the 2D electrical network of variable capacitors, we can obtain very unusual and tunable dispersion diagrams. We present analytic results describing dispersion equations for surface and volume acousto-electric waves and show the corresponding wave simulations. The talk is based on the papers: 1) A. A. Kutsenko, A. L. Shuvalov, and O. Poncelet, "Dispersion spectrum of acoustoelectric waves in 1D piezoelectric crystal coupled with 2D infinite network of capacitors," *J. Appl. Phys.* **123**, 044902 (2018); 2) A. A. Kutsenko, A. L. Shuvalov, O. Poncelet, and A. N. Darinskii, "Tunable effective constants of the one-dimensional piezoelectric phononic crystal with internal connected electrodes," *J. Acoust. Soc. Am.* **137**(2), 606–616 (2015).

Contributed Papers

9:50

3aPA5. Acoustic bianisotropic metasurfaces for broadband non-reciprocal sound transport. Bogdan Ioan Popa, Yuxin Zhai, and HyungSuk Kwon (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, bipopa@umich.edu)

Acoustic bianisotropic materials are media in which the unconventional momentum—strain and stress—particle velocity couplings offer rich mechanical wave dynamics. For instance, transformation elastodynamics have revealed that bianisotropic media could enable extraordinary control over the propagation of elastic waves including the ability to cloak regions of space from detection with mechanical waves. The development of metamaterials have provided the tools to implement these unconventional materials and the first experimental acoustic bianisotropic materials have featured interesting properties such as independent engineering of transmission and reflection. However, despite recent work in this area, the wave dynamics enabled by mechanical bianisotropy have not been fully uncovered and explored. In this presentation we demonstrate experimentally the power of active bianisotropic metasurfaces to produce highly non-reciprocal sound transport in free space in a remarkably broadband fashion and in very compact materials. In addition, we show that this unusual mechanism for non-reciprocal sound propagation is truly linear and does not rely on any type of frequency conversion or other non-linear processes.

10:05–10:20 Break

10:20

3aPA6. Acoustic scattering from a radially-polarized Willis-coupled cylinder. Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Michael.B.Muhlestein@usace.army.mil), Andrew J. Lawrence, Benjamin M. Goldsberry, and Michael R. Haberman (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This talk presents analytical and numerical solutions to the scattering of plane waves from a Willis cylinder with radial polarization. Assuming weak coupling, the scattered field may be explicitly written as a power series expansion of the coupling strength. Using an appropriate choice of average mass density, bulk modulus, and Willis coupling, it is found that the first three scattering modes may be suppressed, providing partial acoustic cloaking using a scattering cancellation approach similar the work of Guild *et al.* [*Phys. Rev. B*, **86**, 104302 (2012).]. Furthermore, we find that the Willis coupling does not need to be strong in order to achieve this mode suppression and so the weak-coupling formulation is appropriate. The scattering problem is then solved numerically using finite element analysis using the weak form of the governing equations, which is then compared to the approximate analytical model for validation. [Work supported by U.S. Army ERDC Geospatial Research and Engineering business area and by ONR.]

10:35

3aPA7. Guided waves at bianisotropic fluid interfaces. Samuel P. Wallen (Appl. Res. Labs. - The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, walles@uw.edu), Caleb F. Sieck (NRC Postdoctoral Res. Associate Program, U.S. Naval Res. Lab., Washington, DC), Benjamin M. Goldsberry (Mech. Eng., The Univ. of Texas at Austin, Austin, TX), Matthew D. Guild, Gregory Orris (U.S. Naval Res. Lab., Washington, DC), and Michael R. Haberman (Appl. Res. Labs. - The Univ. of Texas at Austin, Austin, TX)

Willis fluids are characterized by constitutive relations that couple the pressure and momentum density to both the particle velocity and the volume strain. This effective dynamic response coupling may arise due to microstructural asymmetry, long range order, or time-varying material properties and has been shown to be analogous to electromagnetic bianisotropic media [Phys. Rev. B **96**, 104303 (2017)]. In this study, we report on the existence of guided waves at the interface between two fluids when at least one displays Willis coupling. Criteria for the existence of these waves are discussed in terms of the material properties, frequency, and wave number, and expressions for the dispersion relation and rate of spatial decay away from the interface are obtained analytically. We demonstrate that interface waves are supported when one of the fluids possesses Willis coupling, in contrast to an interface between two classical isotropic fluids, which cannot support interface waves. Special cases are highlighted via numerical examples. [Work supported by ONR, NSF, the Applied Research Laboratories at The University of Texas at Austin, and the NRC Research Associateship Program.]

10:50

3aPA8. Acoustic scattering by a Willis-coupled fluid cylinder. Feruza Amirkulova (Mech. Eng., San Jose State Univ., 1215 Wilbraham Rd., Springfield, MA 01119, feruza@scarletmail.rutgers.edu) and Andrew Norris (Mech. and Aerosp. Eng., Rutgers The State Univ. of New Jersey, Piscataway, NJ)

We present an integral equation based approach to analyze an acoustic scattering from fluid cylinders exhibiting Willis coupling. The Willis-coupled cylinder is characterized by a bulk modulus and mass density as well as a coupling vector. The coupling vector relates the pressure and momentum density to the volume strain and particle velocity simultaneously. We consider integral representations obtained using the free-space Green's function for the exterior fluid medium. The integral equations are evaluated using the T-matrix method. The computation of scattered pressure wave due to an obliquely incident wave on bi-anisotropic fluid cylinder immersed in an external homogeneous fluid will be described. The method will be illustrated giving examples for fluid Willis cylinders.

11:05

3aPA9. Acoustic bianisotropy in effective fluids of finite domain. Caleb F. Sieck, Matthew D. Guild (Code 7160, NRC Postdoctoral Res. Associateship Program, U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, caleb.sieck.ctr@nrl.navy.mil), Andrea Alù (Dept. of Phys. and Elec. Eng., City Univ. of New York, New York, NY), Michael R. Haberman (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Gregory Orris (Code 7160, U.S. Naval Res. Lab., Washington, DC)

Recent research has shown that subwavelength asymmetry and/or nonlocal effects in heterogeneous acoustic media can be described as a dynamic effective fluid displaying acoustic bianisotropic properties. Theoretical homogenization schemes have demonstrated emergent acoustic bianisotropy for an infinite array of subwavelength inhomogeneities in a fluid matrix, and experimental studies have demonstrated individual asymmetric microstructures and single layer metasurfaces exhibiting bianisotropy. However, metamaterials research has shown that a finite domain effective fluid behaves unlike an infinite domain or a surface, and the effective properties of the domain tend to depend on the chosen boundaries and on the position within the domain, even when the microstructure is periodic. This work presents the use of multiple scattering homogenization to discuss the dependence of bianisotropic properties of finite effective fluids on the choice of boundaries and domain size in order to inform future experimental studies on acoustic bianisotropy. [Work supported by the Office of Naval Research and the NRC Research Associateship Program.]

11:20

3aPA10. Acoustic focusing via guided waves in bianisotropic fluid layers. Samuel P. Wallen (Appl. Res. Labs. - The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, walles@uw.edu), Andrew J. Lawrence, Benjamin M. Goldsberry (Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Michael R. Haberman (Appl. Res. Labs. - The Univ. of Texas at Austin, Austin, TX)

Willis fluids display a complex, effective dynamic response stemming from microstructural asymmetry, long range order, and/or time-varying material properties. These materials are characterized by constitutive relations that couple the pressure and momentum density to both the particle velocity and the volume strain. This coupling has been shown to be analogous to electromagnetic bianisotropic media, which exhibit coupled electric and magnetic fields [Phys. Rev. B **96**, 104303 (2017)]. In a recent study, an acoustic lens composed of an array of Willis fluid layers was demonstrated, where the pressure phase imparted by each array element was tuned by varying the direction of Willis polarization and was determined using finite element analysis. In this work, we report on analytical solutions for guided waves in layers of fluid displaying Willis coupling and derive dispersion relations for layers with rigid and pressure-release boundary conditions. Special cases highlighting the effect of Willis polarization direction are presented via numerical examples. The acoustic lens is revisited with array phases obtained from analytical solutions, eliminating the need for finite element methods. Lenses designed for focusing and steering incident pressure waves are demonstrated using fully analytical models. [Work supported by NSF, ONR, and ARL:UT.]

Session 3aPP

Psychological and Physiological Acoustics: Acoustics Bricolage (Poster Session)

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

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

3aPP1. Rippled spectrum resolution in normal listeners: Estimates at different experimental paradigms. Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru), Olga Milekhina, and Dmitry Nechaev (Inst. of Ecology and Evolution, Moscow, Russian Federation)

Rippled-spectrum (RS) signals are used for evaluation of resolution of spectro-temporal structure of sounds. Measurements of spectrum-pattern resolution imply discrimination between the test and reference signals. Therefore, estimates of rippled-pattern resolution may depend not only on the test signal but also on the reference signals. In the present study, the ripples density resolution was measured in combinations of the test and reference signals: (i) RS with phase reversals vs RS without phase reversion; (ii) RS with phase reversals vs non-rippled spectrum; (iii) RS without phase reversals vs non-rippled spectrum; and (iv) RS without phase reversals vs RS of opposite ripple phase. The spectra were centered at 2 kHz and had ERB of 1 oct and SPL of 70 dB. A three-alternative forced-choice procedure was combined with adaptive stimulation procedure. With rippled reference signals, mean ripple resolution limits were 8.9 ripple/oct (phase-reversals test signal) or 7.7 ripple/oct (constant-phase test signal). With non-rippled reference signal, mean resolution limits were 26.1 ripple/oct (phase-reversals test signal) or 22.2 ripple/oct (constant-phase test signal). Different contribution of excitation-pattern and temporal-processing mechanisms is supposed for measurements with rippled and non-rippled reference signals. [Work supported by Russian Science Foundation Grant 16-15-10046.]

3aPP2. Discrimination of band-limited rippled spectra of various central frequencies in cochlear implant users. Dmitry Nechaev (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, dm.nechaev@yandex.ru), Marina Goykhburg, Vigen Bakhshinyan (National Res. Ctr. for Audiol. and Hearing Rehabilitation, Moscow, Russian Federation), Alexander Supin (Inst. of Ecology and Evolution, Moscow, Russian Federation), and George Tavartkiladze (National Res. Ctr. for Audiol. and Hearing Rehabilitation, Moscow, Russian Federation)

Sound signals with wide-band rippled spectra were used for the frequency resolving power measurements in cochlear implant users. In the present study, resolution of band-limited ripple spectra of various central frequencies and speech discrimination test (Russian language) were compared. The rippled spectra had 2-oct cosine envelope with ripples equally spaced on the logarithmic scale. The central frequencies of the rippled spectra were 1, 2, or 4 kHz. Ripple resolution was measured with a three-alternative forced-choice procedure using a paradigm of discrimination of a test signal with ripple phase reversals from a reference signals with a constant-phase ripples. The average ripple discrimination thresholds were 1.9 ripples/octave (RPO) for 1 kHz; 2.3 for 2 kHz; and 2.3 for 4 kHz. The lowest correlation between ripple discrimination limits and speech discrimination results was about 0.2 at a central frequency of 4 kHz, the highest correlation was 0.6 at 1 kHz. In all the cases, the relation between the ripple spectrum resolution and speech discrimination test had positive trends. Band-limited rippled-spectrum tests allow to reveal informative frequency bands for speech discrimination in cochlear implant users. [Work supported by the Russian Science Foundation, Grant No. 16-15-10046.]

3aPP3. Perceiving phonemes with simulated variable cochlear implant insertion depths. Michael L. Smith (Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Box 354875, Seattle, WA 98105, smithm59@uw.edu) and Matthew Winn (Speech Lang. & Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

Cochlear implant (CI) patients have difficulty encoding spectral aspects of a speech signal, and are at a severe disadvantage compared to normal hearing listeners in distinguishing consonants that differ by place of articulation (PoA). One reason for this difficulty could be due to a shifting of spectral energy that occurs with a shallow insertion depth of the electrode array. The present study aimed to measure the effects of spectral shifting on phoneme categorization and PoA perception specifically. Speech was vocoded and then shifted by 0, 2, 4, and 6 mm in cochlear space to simulate shallow CI insertion depth in normal hearing listeners. Stimuli included a /b-d/ and /s-ʃ/ continua that differed by PoA, as well as /ra/ and /la/, to create a six-alternative forced choice paradigm. Results show phoneme categorization became less categorical, with a biased perception of /b-d/ and /s-ʃ/ continuum towards /d/ and /s/ respectively, consistent with higher-frequency energy in those consonants. Some participants were better at recalibrating to the spectrally shifted stimuli than others, as their perception of each continua remained somewhat balanced with increased amounts of spectral shifting. This study showcases the potential consequences and individual differences resulting from shallow insertion depth of a CI.

3aPP4. Cochlear wave propagation under acoustical and electrical stimulations. Amir Nankali (ME Dept., Univ. of Michigan, Ann Arbor, MI 48105, nankali@umich.edu)

Input sounds to the mammalian ear produce waves that travel from the base to the apex of the cochlea. These traveling waves stimulate the microstructure of the organ of Corti (OoC) in the center of the cochlea. The interplay between the traveling wave and a distinct nonlinear amplification process boosts the cochlear responses and enables sound processing over a broad frequency and intensity ranges. The *in vivo* electrical and acoustical stimulations have extensively been used to identify the underlying amplification process. In order to interpret the experimental data and propose a precise active mechanism, a comprehensive three-dimensional model of the cochlea is used. Our model predicts that applying an intracochlear excitation (electrical current or mechanical force) on a location along the cochlea generates a dispersive wave propagation in both forward and reverse directions. It is also found that the wave propagation in the cochlea is nonreciprocal

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when stimulated by an electrical current applied in the OoC. This effect is attributed to the feed-forward mechanism, mediated by a longitudinal electrical cable in our model. [This work was supported by NIH Grant Nos. DC-004084 and T32DC-00011.]

3aPP5. Alleviated cochlear damage in an inflammation suppressing model. Hongzhe Li, Liana Sargsyan, and Alisa Hetrick (VA Loma Linda Healthcare System, 11201 Benton St., Res. Service, Loma Linda, CA 92357, Hongzhe.Li@va.gov)

Cochlear inflammatory response to various environmental insults, has been increasingly become a topic of interest. As the immune response is associated with both pathology and protection, targeting specific components of the immune response is expected to dissect the relationships between cellular damage and inflammation-associated protection and repair in the cochlea. Duffy antigen receptor for chemokines (DARC) is a member of a group of atypical chemokine receptors, and essential for chemokine-regulated leukocyte/neutrophil trafficking during inflammation. Previous studies have reported that *Darc* deficiency alters chemokine bioavailability and leukocyte homeostasis, leading to significant anti-inflammatory effects in tissues following injury. In this study, we have used *Darc* knockout mice to determine the impact of a deficiency in this gene on cochlear development, as well as function in cochlea subjected to various stresses. We observed that DARC is not required for normal development of cochlear function, as evidenced by typical hearing sensitivity in juvenile *Darc*-KO mice, as compared to wild type *C57BL/6* mice. However, *Darc*-KO mice exhibited improved hearing recovery after intense noise exposure when compared to wild-type. At cochlear locations above frequency range of the energy band of damaging noise, both hair cell survival and ribbon synapse density were improved in *Darc* deficient animals.

3aPP6. Comparison of wideband and clinical acoustic reflex thresholds in patients with normal hearing and sensorineural hearing loss. M. P. Feeney (National Ctr. for Rehabilitative Auditory Res., Portland, OR), Kim Schairer (James H. Quillen VAMC, PO Box 4000, Mountain Home, TN 37684, kim.schairer@va.gov), Douglas H. Keefe, Denis Fitzpatrick (Boys Town National Res. Hospital, Omaha, NE), Daniel Putterman, Angie Garinis, Michael Kurth (National Ctr. for Rehabilitative Auditory Res., Portland, OR), Elizabeth Kolberg, Kara McGregor (James H. Quillen VAMC, Mountain Home, TN), and Ashley Light (East Tennessee State Univ., Johnson City, TN)

Acoustic reflex thresholds (ARTs) obtained with a wideband (WB) probe and an adaptive threshold detection procedure were compared to ARTs using a clinical system. Ipsilateral and contralateral ARTs were elicited in a group of 79 adults with normal hearing (NH) and 51 adults with sensorineural hearing loss (SNHL). ARTs were obtained for both methods using activator tones of 0.5, 1.0, and 2.0 kHz and broadband noise (BBN) with a bandwidth extending to 4.0 kHz for the clinical and 8.0 kHz for the WB ART. Results were similar for ipsilateral and contralateral ARTs. Tonal ARTs with the clinical method were slightly elevated for the SNHL group for all three activator tones, but for the WB method were elevated at 2.0 kHz where the average hearing loss was greatest (47 dB HL). ARTs for BBN were higher for the clinical method than the WB method for both groups, and the difference between groups was around 5 dB for the clinical method but 12 dB for the WB method. This suggests that ARTs with the WB method and BBN activator extending to 8 kHz are a more sensitive indicator of high-frequency SNHL than the clinical method. Individual reflex patterns will also be presented.

3aPP7. Wideband transient otoacoustic emissions in ears with normal hearing and sensorineural hearing loss. Kim Schairer (James H. Quillen VAMC, PO Box 4000, Mountain Home, TN 37684, kim.schairer@va.gov), Douglas H. Keefe, Denis Fitzpatrick (Boys Town National Res. Hospital, Omaha, NE), Daniel Putterman (National Ctr. for Rehabilitative Auditory Res., Portland, OR), Elizabeth Kolberg (James H. Quillen VAMC, Mountain Home, TN), Angie Garinis, Michael Kurth (National Ctr. for Rehabilitative Auditory Res., Portland, OR), Kara McGregor (James H. Quillen VAMC, Mountain Home, TN), Ashley Light (East Tennessee State Univ., Johnson City, TN), and M. P. Feeney (National Ctr. for Rehabilitative Auditory Res., Portland, OR)

Otoacoustic emissions (OAEs) are generated in the cochlea in response to sound and are used clinically to separate ears with normal hearing from sensorineural hearing loss (SNHL). OAEs were elicited at ambient pressure by clicks (CEOAE) and wideband chirps (TEOAE) sweeping from low-to-high frequency with a sweep rate of either 187.6 Hz/ms (short chirps) or 58.2 Hz/ms (long chirps) and a bandwidth extending to 8 kHz. Chirps were presented at the same sound exposure level (SEL) as clicks, or + 6 dB relative to clicks. A total of 288 OAE waveforms were averaged for short chirps in ~1 minute compared to 120 waveforms for long chirps. Compared to clicks, the chirp has a lower crest factor, which allows it to be presented at an overall higher SEL without distortion. OAEs were elicited in 79 adults with normal hearing and 51 adults with mild-to-moderate SNHL. One-sixth octave OAE signal-to-noise ratios from 0.7 to 8.0 kHz were compared across stimulus types and conditions. The area under the receiver operating curve (AUC) was used to assess the accuracy of detecting SNHL. Average AUCs across 1/6th octave frequencies ranged from 0.90 to 0.89 for TEOAEs and were 0.87 for the CEOAE suggesting excellent test performance.

3aPP8. Behavioral methodology demonstrating the development of Tinnitus from chronic non-traumatic noise in mice. Kali Burke, Laurel A. Screven (Psych., Univ. at Buffalo, SUNY, 246 Park Hall, Buffalo, NY 14260, kaliburk@buffalo.edu), Matthew A. Xu-Friedman (Biological Sci., Univ. at Buffalo, SUNY, Buffalo, NY), and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, Buffalo, NY)

Tinnitus is the perception of a ringing or hissing noise in the absence of an external auditory stimulus. It can develop in one or both ears following acute or chronic noise exposure, head trauma, or as the result of natural aging. Using the Stolzberg *et al.* (2013, *J. Neurosci. Methods*) identification paradigm for rats, we assessed tinnitus in awake and behaving mice in response to long-term, non-traumatic noise exposure. Mice were trained to categorize three stimuli into two categories. Category 1 was 1/8 octave narrow band noise bursts centered around 4, 8, 16, 22.6, and 32 kHz. Category 2 was amplitude modulated white noise (modulated 100% at a rate of 5 kHz) and silent trials. Following this training period, mice lived in 85 dB noise for 23 hours a day and were tested for the remaining 1-hour each day. Most of the mice developed tinnitus, which was indicated by a shift in the categorization of the silence trials from category 2 to category 1. The present experiment demonstrates that chronic, non-traumatic noise exposure can induce tinnitus in mice in a similar way to salicylate injections in rats.

3aPP9. The influence of time of hearing aid use on auditory perception in various acoustic situations. Piotr Szymanski (Training and Development Dept., GEERS Hearing Acoustics, Lodz, Poland), Tomasz Poremski (Audiol. Support Ctr., GEERS Hearing Acoustics , Lodz, Poland), and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Faculty of ETI, Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

The assessment of sound perception in hearing aids, especially in the context of benefits that a prosthesis can bring, is a complex issue. The objective parameters of the hearing aids can easily be determined. These

parameters, however, do not always have a direct and decisive influence on the subjective assessment of quality of the patient's hearing while using a hearing aid. The paper presents the development of a method for the assessment of auditory perception and the effectiveness of applying hearing aids for hearing-impaired people during a short-term use. The method involves a questionnaire based on the APHAB (Abbreviated Profile of Hearing Aid Benefit) assessment questionnaire, a measure of self-reported auditory disability. The study includes additional criteria, such as measuring the number of hours and days of use of hearing aids, the degree of hearing loss and the patient's experience. A web-based application is developed to enable to carry out such an examination from any computer with access to the network. The research results show that in the first period of use of hearing aids, speech perception improves, especially in noisy environments. The perception of unpleasant sounds also increases, which leads to deterioration of hearing aid acceptance by their users.

3aPP10. Development and calibration of a smartphone application for use in sound mapping. Lawrence L. Feth, Evelyn M. Hoglund, Gus Workman, Jared Williams, Morgan Raney, and Megan Phillips (Speech and Hearing Sci., Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, feth.1@osu.edu)

Klyn and Feth (2016) reported preliminary work to use a smartphone application in a citizen-science project designed to map sound levels in Columbus, OH. Before the main project began, we discovered that the sound level measuring applications available for download had shortcomings that made them unsuitable for the proposed work. This presentation describes the development of two smartphone applications, one iOS and one Android, and the calibration procedures developed to document their accuracy and reliability. Following the suggestions of Kardous, *et al.* (2014, 2016), we require that the measurements be conducted using an external microphone. In use, microphone voltage is sampled for 30 seconds and processed to reflect the A-weighting scale so that sound levels are recorded as dBA values. The time and location of each sample is saved with the sound level value and can only be uploaded to the project data base if the device has been previously calibrated. Calibration stores an offset value that can be added to each sample before producing mapped values. Limits on proximity of repeated samples are included to ensure a better distribution of results. The apps are currently available for download from their respective "stores." [Work supported by a grant from the Battelle Engineering, Technology and Human Affairs Endowment.]

3aPP11. The role of pitch and harmonic cancellation when listening to speech in background sounds. Daniel Guest and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 E River Rd., Minneapolis, MN 55455, guest121@umn.edu)

Fundamental frequency (F0) differences between competing talkers can aid their perceptual segregation (ΔF0 benefit), but the underlying mechanisms remain poorly understood. One theory of ΔF0 benefit, harmonic cancellation, proposes that the periodicity of the masker can be used to cancel (i.e., filter out) its neural representation. Earlier work suggested that a one-octave ΔF0 provided little benefit, an effect predicted by harmonic cancellation due to the shared periodicity of the masker and target. An alternative explanation is that this effect is simply due to spectral overlap between the target and masker. To assess these competing explanations, speech intelligibility of a monotonized target talker masked by a speech-shaped harmonic complex tone was measured as a function of ΔF0 (0, +3, +12, +15 semitones) and spectral structure of the masker (all harmonics, odd harmonics only). We found that removal of the masker's even harmonics when the target F0 was one octave higher than the masker improved speech reception thresholds by about 6 dB. Because this manipulation eliminated spectral overlap between target and masker components but did not alter their shared periodicity, the finding is consistent with the explanation based on spectral overlap, but not cancellation. [Work supported by NIH R01DC005216 and NSF NRT-UtB1734815 grants.]

3aPP12. Acoustic features in speech for emergency perception. Maori Kobayashi (Japan Adv. Inst. of Sci. and Technol., 1-1, Asahidai, Nomi, Ishikawa 9231292, Japan, maori-k@jaist.ac.jp), Yasuhiro Hamada (Japan Adv. Inst. of Sci. and Technol., Nakano, Japan), and Masato Akagi (Japan Adv. Inst. of Sci. and Technol., Nomi, Ishikawa, Japan)

Previous studies have reported that the acoustic features such as the speech rate, fundamental frequency (F0), amplitude, and voice gender are related to emergency perception in speech. However, the most critical factor influencing the emergency perception in speech remains unknown. In this study, we compared influences of three acoustic features (speech rate, F0, and spectral sequence (amplitude)) to determine the acoustic feature that has the most influence on emergency perception in speech. Prior to conducting our experiments, we selected five speech phrases with different level of perceived emergency among various speech phrase spoken by TV casters during real emergencies. We then created synthesized voices by replacing three acoustic features separately among the selected five voices. In experiment 1, we presented these synthesized voices to 10 participants and asked them to evaluate levels of the perceived emergency of each voice by the magnitude estimation method. The results from experiment 1 showed that F0 was most influential on emergency perception. In experiment 2, we examined influences of the three acoustic features on auditory impression related to the perceived emergency by the SD method. The results suggested that emotional effects of some words such as "tense" or/and "rush" were influenced by the fundamental frequency.

3aPP13. Irrelevant sound effects with locally time-reversed speech: Native vs. non-native language. Kazuo Ueda, Yoshitaka Nakajima (Human Sci., Kyushu Univ., 4-9-1 Shiobaru, Minami-ku, Fukuoka 815-8540, Japan, ueda@design.kyushu-u.ac.jp), Wolfgang Ellermeier (Technische Universitaete Darmstadt, Darmstadt, Germany), and Florian Kattner (Technische Universitaete Darmstadt, Darnstadt, Germany)

To disentangle the contributions of local and global temporal characteristics of irrelevant speech in native and non-native language, irrelevant speech/sound effects (ISEs) of normal speech, reversed speech, locally time-reversed speech [Ueda *et al.* (2017). Sci. Rep. 7:1782] and its reversal on serial-recall of visually presented series of digits were examined. ISE experiments were performed with German native listeners (n= 79) and with Japanese native listeners (n= 81), employing both German and Japanese speech with either sample. All conditions involving speech significantly impaired memory performance when compared to a pink-noise control condition. When the native language of each group of listeners was presented, locally time-reversed speech with the shortest segment duration (20 ms), which was highly intelligible, was as equally disruptive as normal speech to the memory task, whereas locally time-reversed speech with longer segment durations (70 or 120 ms) was less disruptive. The effect of segment duration totally disappeared when the locally time-reversed speech was played backward. When the non-native language was presented, locally time-reversed speech showed the same pattern of results in each language group, irrespective of the direction of playback. Thus, the irrelevant sound effect worked differently for one's native and non-native languages.

3aPP14. Low-rate frequency modulation detection across the lifespan. John Grose and Emily Buss (Univ. of North Carolina at Chapel Hill, Dept. OHNS, CB#7070, 170 Manning Dr., Chapel Hill, NC 27599, john_grose@med.unc.edu)

Frequency modulation (FM) detection at a 2-Hz rate is thought to rely on temporal cues. For a 500-Hz carrier, this reliance is heightened by the use of dichotic presentation. Here, the modulator is inverted across ears such that binaural beats can cue FM. Detection acuity for FM is better in the dichotic than diotic condition, promoting the task as a measure of temporal fine structure (TFS) sensitivity. Older adults show less benefit of dichotic presentation than young adults, suggesting deficient TFS processing. This study extended the work to school-age children in order to gain a

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perspective of TFS processing across the lifespan. Children aged 4–10 yrs showed poorer FM detection in both diotic and dichotic conditions re young adults, and their performance was more similar to that of older adults. The dependence of the task on TFS processing was further tested by adding background noise. This reduced FM detection acuity but more for the dichotic than diotic condition. Overall, these results support the dependence of the FM detection task on TFS processing and demonstrate that such processing appears compromised at both ends of the age spectrum. [Work supported by NIDCD DC01507 (JHG) and DC00397 [EB].)

3aPP15. Loudness constancy in healthy older adults: Effects of sound production and visual cues of music playing. Akio Honda (Shizuoka Inst. of Sci. and Technol., 2200-2 Toyosawa, Fukuroi, Shizuoka 437-8555, Japan, akio.honda6@gmail.com), Ayumi Yasukouchi, and Yoichi Sugita (Waseda Univ., Tokyo, Japan)

Loudness constancy refers to the phenomenon by which loudness remains constant in the presence of substantial changes in a physical stimulus caused by varying the sound distance. We investigated loudness constancy in healthy older adults. The degree of loudness constancy was measured using two methods of adjustment: "sound production," by which listeners played a musical instrument as loudly as a model player, and "sound level adjustment," by which listeners adjusted the loudness of the sound produced by a loudspeaker. The target sound was produced by the actual musical instrument performance. Sound pressure levels of the stimuli were approximately 60, 75, and 86 dB(A). The distances between the performer and the participant were 2, 8, and 32 m. In both conditions, participants were asked to produce the level of sound pressure matching the stimulus. Results show that when visual cues of musical performance are available, sound production had more robust loudness constancy than the sound level adjustment method. These results support an earlier claim that audiovisual perception and imitation are necessary for musical learning and skill acquisition.

3aPP16. Bilingual speech associations in phonetic representation. Margaret Cychosz (Dept. of Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720, mcychosz@berkeley.edu) and Erik C. Tracy (Psych., Univ. of North Carolina at Pembroke, Pembroke, NC)

Listeners unconsciously index race, gender, and sexual orientation from little phonetic input. Psycholinguistic theory attributes this to a representation of language where episodic traces encode fine, speaker-specific acoustic detail. We examined this unconscious linguistic bias in a task with native English speaker participants. Participants were presented with two pictures, both of which were either Hispanic or Caucasian males. Simultaneously, they listened to a semantically-neutral English sentence spoken by either a bilingual Spanish-English male or a monolingual English male and were told to choose which pictured man said the sentence. When the auditory stimulus was bilingual speech, speakers were quicker to choose between the two Hispanic faces than between the Caucasian faces. However, when participants heard a monolingual voice, there was no difference in reaction time. In the second phase of the experiment, participants were instead presented with two different pictures, one Hispanic and one Caucasian. Listeners were more likely to associate bilingual English with the Hispanic male than the Caucasian male and vice versa (monolingual English with Caucasian male). These results suggest that listeners quickly index male Spanish-English speech, but monolingual English-Caucasian associations may not be as robust.

3aPP17. Notches in sentence spectra bias subsequent phoneme categorization. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Spectral differences across earlier (context) and later (target) sounds are perceptually magnified, resulting in spectral contrast effects (SCEs) that bias categorization of later sounds. Most studies added spectral peaks to context sounds in order to produce SCEs, but noise contexts with spectral notches at speech frequencies biased phoneme categorization in a complementary manner (Coady *et al*., 2003 *JASA*). We tested whether this approach generalized to speech contexts with spectral notches. On each trial, a context sentence preceded the target phoneme (Experiment 1: /ɪ/-/ɛ/; Experiment 2: /d/-/g/). Sentences were processed by notch filters that attenuated energy in the same frequency regions that, when amplified to add spectral peaks, produced SCEs (Experiment 1: 100–400/550–850 Hz, as in Stilp *et al*., 2015 *JASA*; Experiment 2: 1700–2700/2700–3700 Hz, as in Stilp & Assgari, 2017 *JASA*). Notch depths ranged from −5 to −20 dB in 5-dB steps. In both experiments, notch-filtered sentences biased phoneme categorization in complementary directions to SCEs, with bias magnitudes increasing at larger notch depths. Whether earlier sounds have spectral peaks or notches, speech categorization is highly sensitive to the magnitudes of spectral differences across context and target sounds.

3aPP18. Quasi-phonemic vowels in Ampenan Sasak: An acoustic analysis. Leah Pappas (Linguist, Univ. of Hawai'i at Mānoa, 1890 East-West Rd., Moore Hall, Honolulu, HI 96822, lpappas@hawaii.edu)

Phonological studies on Sasak tend to agree that Sasak has a six-vowel system /i, u, e, o, ə, a/ (Archangeli *et al*., n.d.; Chahal, 1998; Jacq, 1998). However, in Ampenan Sasak, a dialect spoken in the provincial capital of Mataram, it is difficult to attribute vowel quality differences of tense mid-vowels [e, o] and lax mid-vowels [ɛ, ɔ] to allophonic variation of underlying phonemes /e, o/. While lax mid-vowels tend to appear in heavy syllables and tense mid-vowels appear in light syllables (see also Archangeli *et al*., n.d.), there are exceptions (e.g., /baɾeh/ "late," /əmbong/ "reservoir") and several minimal pairs (e.g., /bəɾəmbok̚/ "discuss," /bəɾəmbɔk̚/ "breathe;" /koboʔ/ "levening," /kɔbɔʔ/ "play with water"). Further, speakers' intuitions about these distinctions are unreliable. In order to understand this relationship, we elicited wordlists across 13 female speakers of Ampenan Sasak aged 18 to 48. Wordlists were balanced for syllable weight, coda segment, and syllable position and subjected to acoustic analysis. Preliminary results show that the quality of the vowel is highly affected by the weight of the syllable and the coda segment. This suggests that mid-vowels are quasi-phonemic in Ampenan Sasak (Ladd 2013). While they generally behave as allophones, this paradigm is not strictly followed.

3aPP19. Effects of incomplete feedback on response bias in auditory detection reanalyzed. Shuang Liu (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, Columbus, OH 43210, liu.3267@osu.edu), Matthew Davis (CDO Technologies, Inc., Columbus, OH), and Lawrence L. Feth (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Davis (2015) reported the effects of providing incomplete feedback to listeners in a simple detection task. When feedback is limited in a detection experiment, the observer's response criterion may deviate significantly from the optimal criterion. To approximate real-world listening conditions, a single-interval yes-no procedure was conducted with 10 feedback conditions ranging from no feedback to complete feedback. The signal was a brief 1 kHz tone; the masker was wideband white noise. The SIAM procedure (Kaernbach, 1990) was used to establish the SNR for 75% detection threshold for each listener. That level was then used in each of the feedback conditions. Davis reported a detailed descriptive analysis of the symmetry, organization, implicitness, and amount of feedback and the individual differences noted across the ten listeners. For this project, an equal-variance Gaussian signal detection framework was used to analyze the data. Model parameters were estimated via Bayesian inference. The main finding is that, as expected, complete feedback drives response criteria toward the optimum, and deviation from the optimal criterion increases as the amount of feedback decreases. While most subjects show this general trend, a few subjects maintain near-optimal behavior throughout all conditions, which is not a surprise.

3aPP20. The effect of experience and tonal contour: A preliminary study of perceived duration on Mandarin tones. Yu-an Lu, Yu-Ming Chang (Foreign Lang. & Literatures, National Chiao Tung Univ., Taiwan, F319, Humanities Bldg. 2, 1001 University Rd., Hsinchu 30010, Taiwan, yuanlu@nctu.edu.tw), Yang-Yu Chen, Peichun Chen, Shao-Jie Jin, Cheng-Huan Lee, Waan-Rur Lu, Yen-Ju Lu, and Yu Nan (Foreign Lang. & Literatures, National Chiao Tung Univ., Taiwan, Hsinchu, Taiwan, Taiwan)

Mandarin tones are shown to be produced with different lengths (i.e., from longer to shorter: T3 > T2 > T1 > T4) (cf. Wu & Kenstowicz, 2015).

An AX rating experiment in which Taiwan Mandarin listeners were asked to rate the relative durations of syllables ([pa], [pi], [ta], [ti]) manipulated into five different duration steps (290 ms, 320 ms, 350 ms, 380 ms, and 410 ms) in Mandarin tones (high-level T1, rising T2, dipping T3, reduced low-level T3, and falling T4) compared with an anchor stimulus ([pa] with 350 ms in mid-level tone) showed that the complex contour tone (T3) was rated as longer than simple contour tones (T2 and T4) and simple contour tones were rated as longer than level tones (T1 and reduced T3). Between the simple contour tones, T2 was rated as longer than T4. Between the level tones, the reduced T3 was rated as longer than T1. The explanations to these tonal perceptual differences are tied to the typological correlation between rime duration and the complexity of tonal targets (T3 > T2/T4 > T1/T3) (e.g., Zhang 2001) as well as to the listeners' experience to the durations of different tones (T2 > T4 and reduced T3 > T1).

3aPP21. Acoustic and articulatory evidence on incomplete syllable-final nasal mergers in Taiwan Mandarin. Chenhao Chiu (Graduate Inst. of Linguist, National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, chenhaochiu@ntu.edu.tw), Yu-an Lu (Foreign Lang. and Literatures, National Chiao Tung Univ., Hsinchu, Taiwan), and Yining Weng (Graduate Inst. of Linguist, National Taiwan Univ., Taipei, Taiwan)

Previous studies reported that syllable-final nasals in Taiwan Mandarin, /n/ and /ŋ/, exhibit merging to different degrees in the vowel contexts of /i/ and /ə/ (e.g., Ing 1985, Kubler 1985, Hsu & Tse 2007, Fon *et al*., 2011). However, the discussion has been mainly on the merging directions, ethnic groups that display the mergers, and attitudes towards the different merging directions. Apart from the large evidence of merging from perceptual identification of the final nasal, this study employs ultrasound imaging technique to examine tongue postures of the syllable-final nasal following /i, a, ə/ vowel contexts. Preliminary results showed that /n/ and /ŋ/ were distinct both acoustically and articulatorily in the context of /a/, echoing earlier findings (e.g., Fon *et al*., 2011). On the other hand, while the contours of tongue shapes demonstrate extensive overlapping between /n/ and /ŋ/ following /i/ and /ə/, their correspondent acoustics display different degrees of nasalization on the preceding vowels (cf. Chen, 2000). More generally, the results suggest that the merging in syllable-final nasals undergoes an incomplete process and the acoustic contrasts might have lie in the nasalization of the preceding vowels and less so in syllable-final nasals.

3aPP22. Effects of corrective feedback on receptive skills in non-native contrasts: A training study in Taiwan Southern Min. YinChing Chang (Graduate Inst. of Linguist, National Taiwan Univ., 4F.-E, No.64-1, Sec. 3, Xinsheng S. Rd., Da'an Dist., Taipei City 106, Taipei 10660, Taiwan, ching199541@gmail.com)

The benefits of corrective feedback (CF) have been proposed in numerous L2 speech perception training studies, concerning that it provides learners with opportunities to retrieve, restructure, and consolidate their phonological representations. Moreover, different types of CF have been investigated considering that different effects will be triggered owing to involving different types of cognitive processes. This study examines the role of different types of CF in receptive skill, focusing on voiceless-voiced contrasts in Taiwan Southern Min, which is absent in Mandarin. Native speakers of Taiwan Mandarin who are also non-native listeners of Taiwan Southern Min are randomly assigned to three groups with different types of CF: Target, Non-target, and Combination of target and non-target. Preliminary results showed that among the three CF groups, identification accuracy significantly improved when participants are provided with combination CF. More generally, the results suggest that corrective feedback can play an influential role in identifying phonemic contrasts that may not occur in listeners' native language.

3aPP23. Coordinated systems in Taiwan Mandarin tone production: An investigation of laryngeal and tongue movements. Chenhao Chiu (Graduate Inst. of Linguist, National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Graduate Inst. of Linguist, Taipei 10617, Taiwan, chenhaochiu@ntu.edu.tw)

While electroglottography (EGG) is commonly used to examine different phonation types as well as laryngeal closure and opening of segments, only limited EGG studies focus on tonal quality [Brunelle *et al*., 2010, *Phonetica*, 67:147]. In particular, it is not yet clear how laryngeal movements in terms of its vibration magnitude and vertical displacement may be associated with pitch contour in tone production. The current study tackles this question by examining the amplitude of EGG pulses and laryngeal heights across four tones in Taiwan Mandarin. Ultrasound images are also obtained to assess the temporal relationship between tongue positioning and laryngeal movements during tone production. Preliminary results show that while the laryngeal area is more constricted by low, back vowels, a stronger correlation between EGG amplitude and pitch contour is found. On the other hand, when the tongue is positioned in the front of the cavity, larger degrees of freedom around the laryngeal area yield a lesser correlation between EGG amplitude and pitch contour but a stronger correlation between laryngeal height and pitch contour. The results suggest that tone production may be a result of a combination of laryngeal vibration and spatial coordination between the laryngeal and supralaryngeal movements. [Funding from MOST.]

3aPP24. Frequency-following responses elicited by a consonant-vowel with intonation. Kristin M. Stump and Fuh-Cherng Jeng (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W224, Athens, OH 45701, jeng@ohio.edu)

To date, two major types of acoustic stimuli have been used to evoke frequency-following responses (FFRs)—either by using a consonant-vowel or by using a vowel with intonation, but not both. The goals of this study were to (1) determine the feasibility of recording FFRs by forging a CV and intonation into one acoustic stimulus, and (2) examine the characteristics of such responses. Twelve Chinese adults (8 females, 4 males, age = 25.58 ± 4.23 years old) were recruited. FFRs were elicited by using the consonant-vowel /da/ combined with a rising intonation pattern (fundamental frequencies 85-93 Hz) for a total of 8000 sweeps from each participant. Six objective indices (consonant amplitude, consonant latency, consonant latency, frequency error, tracking accuracy, and pitch strength) were derived from all recordings. Results demonstrated that it was feasible to record the consonant, vowel, and intonation responses simultaneously. Results also demonstrated distinctive FFR trends with increasing number of sweeps for the consonant, vowel, and intonation responses. Taken together, these findings may have important implications for basic research and clinical applications.

3aPP25. Machine learning in detecting frequency-following responses. Breanna Hart and Fuh-Cherng Jeng (Communications Sci. and Disord., Ohio Univ., 366 Richland Ave., Apt. 2303, Athens, OH 45701, bh130015@ohio.edu)

To improve the efficiency and timeliness in frequency-following response (FFR) testing, the purpose of this study was to investigate the capabilities of machine learning in the detection of an FFR. Continuous brain waves were recorded from 25 Chinese adults in response to a pre-recorded Mandarin monosyllable \yi˧˥\ with a rising frequency contour. A total of 8000 artifact-free sweeps were recorded from each participant. Continuous

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brain waves accumulated (from the first sweep) up to the first 500 sweeps were considered FFR absent; brain waves accumulated (from the first sweep) up to the last 1000 sweeps (i.e., from 7001 to 8000 sweeps) were considered FFR present. Six response features (frequency error, slope error, tracking accuracy, spectral amplitude, pitch strength, and root-mean-square amplitude) were extracted from each recording and served as key predictors in the identification of a response. Twenty-three supervised machine-learning algorithms, with a 10-fold cross-validation procedure, were implemented via a Classification Learner App in MATLAB. Two algorithms yielded 100% efficiency (i.e., 100% sensitivity and 100% specificity) and 14 others produced efficiency $\geq$99%. Results indicated that a majority of the machine-learning algorithms provided efficient and accurate predictions in whether an FFR was present or absent in a recording.

3aPP26. Effects of auditory selective attention on word intelligibility and detection threshold of narrow-band noise. Ryo Teraoka, Shuichi Sakamoto, Zhenglie Cui, Yôiti Suzuki, and Satoshi Shioiri (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, terar@dc.tohoku.ac.jp)

By directing auditory selective attention, humans can recognize a target sound from multiple environmental sounds. Evidence suggests that the effect of auditory selective attention is observed not only in the frequency domain, but also in the spatial domain. However, the effect of auditory spatial attention is not always observed and has been suggested to depend on the level of auditory processing. Therefore, the task dependency of auditory spatial attention was investigated by measuring the word intelligibility and detection threshold of target 1/12 octave-band noise amongst those with different center frequencies. Target and distractor sounds were presented by a loudspeaker array surrounding listeners. The target sound was presented after listeners' auditory attention was attracted to the specific direction from which the target sounds would be presented. Results showed that the word intelligibility increased when attended but the detection threshold of narrow-band noises did not change significantly. This suggests that auditory processing higher than that required for detecting a specific random noise among spatially distributed similar noises is necessary for the effect auditory spatial attention to appear. It seems interesting to pursue the lowest level, where the effects of auditory spatial attention are observable.

3aPP27. Hemispheric decoupling of awareness-related activity in human auditory cortex under informational masking and divided attention. Andrew R. Dykstra (The Brain and Mind Inst., Univ. of Western ON, 1151 Richmond St., Western Interdisciplinary Res. Bldg., London, ON N6A 3K7, Canada, andrew.r.dykstra@gmail.com) and Alexander Gutschalk (Dept. of Neurology, Ruprecht-Karls-Universität Heidelberg, Heidelberg, Germany)

Whether unattended sound streams reach perceptual awareness, and the extent to which they are represented in the central auditory system, are fundamental questions of modern hearing science. We examined these questions utilizing M/EEG, multi-tone masking, and a dual-task dichotic listening paradigm. Listeners performed a demanding primary task in one ear—detecting isochronous target-tone streams embedded in random multi-tone backgrounds and counting within-target-stream deviants—and retrospectively reported their awareness of similar masker-embedded targets in the other ear. Irrespective of attention or stimulation ear, left-AC activity strongly covaried with target-stream detection starting as early as 50 ms post-stimulus. In contrast, right-AC activity, while highly sensitive to stimulation ear, was unmodulated by detection until later, and then only weakly. Thus, under certain conditions, human ACs can functionally decouple, such that one—here, right—is automatic and stimulus-driven while the other—here, left—supports perceptual and/or task demands, including basic perceptual awareness of nonverbal sounds both within and outside the focus of top-down selective attention.

3aPP28. Neural correlates of beat perception in vibro-tactile modalities. Sean A. Gilmore, Gabriel Nespoli, and Frank A. Russo (Psych., Ryerson Univ., 1168 Dufferin St., Toronto, ON M6H 4B8, Canada, sean.gilmore@ryerson.ca)

Musical rhythms elicits a perception of a beat (or pulse) which in turn elicit spontaneous motor synchronization (Repp & Su, 2013). Electroencephalography (EEG) research has shown that endogenous neural oscillations dynamically entrain to beat frequencies of musical rhythms providing a neurological marker for beat perception(Nozaradan, Peretz, Missal, & Mouraux, 2011). Rhythms however, vary on in complexity modulating ability to synchronize motor movement. Although musical rhythms are assumed to be from auditory sources, recent research suggests that rhythms presented through vibro-tactile stimulation of the spine elicits a comparable capacity as auditory in motor synchronization in simpler rhythms, however, this trend diminishes as complexity increases(Ammirante, Patel, & Russo, 2016). The current research purposes to explore the neural correlates of vibro-tactile beat perception with the aim in providing further evidence for rhythm perception from a vibro-tactile modality. Participants will be passively exposed to simple and complex rhythms from auditory, vibro-tactile, and multi-modal sources. Synchronization ability as well as EEG recording will be obtained in order to provide behavioural and neurological indexes of beat perception. Results from this research will provide evidence for non-auditory, vibro-tactile capabilities of music perception.

3aPP29. Analysis of spectral and transmission characteristics of bone-conducted speech using real utterances and transcutaneous vibration. Teruki Toya (School of Adv. Sci. and Technol., Japan Adv. Inst. of Sci. and Technol., 1-1, Asahidai, Nomi-shi 923-1292, Japan, yattin_yatson@jaist.ac.jp), Peter Birkholz (Juniorprofessur für Kognitive Systeme, Technische Universität Dresden, Dresden, Sachsen, Germany), and Masashi Unoki (School of Adv. Sci. and Technol., Japan Adv. Inst. of Sci. and Technol., Nomi-shi, Japan)

Previous studies related to auditory feedback have found that hearing one's own voice has important roles in maintaining one's speech production. However, it is still unclear how perception of one's bone-conducted (BC) speech affects one's speech production, since most studies have just presented white/pink noise for masking speakers' BC speech without enough consideration of acoustical and transmission characteristics of BC speech itself. This study aims to investigate spectral characteristics of the observable parts of BC speech (temporal bone (TB) vibration and ear canal (EC) speech) and transmission characteristics related to them. Spectral envelopes of vowel utterances and long-term average spectrum (LTAS) of sentence utterances were analyzed. Moreover, transfer functions from larynxes to TB and EC were measured using transcutaneous vibration signals. As results of these analyses, although the spectral shapes are different among the vowels in TB vibration and EC speech, it was found that the transmission from larynxes to TB emphasizes lower frequency components and the transmission from larynxes to EC emphasizes the higher frequency components. [Work supported by JSPS KAKENHI Grant No. JP. 17J03679, and Grant in Aid for Scientific Research Innovative Areas (Grant No. 16H01669, 18H05004) from MEXT, Japan.]

3aPP30. Cyclic movement primitives underlying two-handed alternating signs in signed language. Oksana Tkachman, Grace Purnomo, and Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, tox.cs84@gmail.com)

Biomechanical constraints have been described as underlying patterns in speech production: e.g., cyclic neural central pattern generators (CPGs) in the jaw that evolved for chewing have been claimed to affect syllabic patterns [MacNeilage 1998, *Behav. Brain Sci.* 21: 4] and cyclic patterns in tongue tip movement may span sequences of segments [Derrick & al.

2015, *JASA-EL* 137:3]. Similarly, we propose that otherwise unexplained universal aspects of sign languages may result from a preference for repeated alternating arm movements developed in human ancestors for quadrupedal locomotion. For example, in ASL-LEX corpus of 1000 ASL signs, 159 are balanced (not including signs without movement or articulated in the transverse plane, which has alternative properties we will address in the talk). Of those, 60% are alternating, and the alternating group had fewer iconic signs (30.5% vs 41%) and more repeated signs (61% vs 30%) than symmetrical signs. While both groups had a large proportion of signs realized on coronal plane, which makes them more visible in face-to-face communication (33% for alternating vs 52% for symmetrical), the former have more signs in midsagittal and coronal/transverse planes (63% vs 44%), as predicted. Supporting data from additional sign languages will be presented.

WEDNESDAY MORNING, 7 NOVEMBER 2018 SALON A (VCC), 8:00 A.M. TO 12:00 NOON

Session 3aSC

Speech Communication, Musical Acoustics, and Psychological and Physiological Acoustics: The Sound of Emotion

Shae D. Morgan, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

3aSC1. The acoustic and neurobiological bases of the processing of vocal emotion. Sophie K. Scott (Inst. for Cognit. Neurosci., UCL, 17 Queen Square, London WC1N 3AZ, United Kingdom, sophie.scott@ucl.ac.uk)

In this talk, I will briefly address the ways that the expression of emotion in the voice is affected by physiological processes (e.g., the effects of adrenaline on the vocal folds), as well an interplay of voluntary and involuntary aspects of vocal control. I will outline some behavioural and functional imaging studies of the perception of emotional voices, and the ways that emotion in the voice can be associated with some general acoustic properties (e.g., duration correlates negatively with ratings of arousal). However there is weaker evidence for a relationship between emotional valence and acoustic factors, as well as considerable evidence for emotion specific relationships between acoustic profiles and emotion ratings. I will extend this into a consideration of the sensori-motor networks recruited by different kinds of emotional vocalisations, and argue that this can often reflect more emotion specific effects, which can be linked to both affective and behavioural aspects of emotional processing.

8:25

3aSC2. The experience of music emotion in older adults with hearing loss and hearing aids. Frank A. Russo (Psych., Ryerson Univ., 350 Victoria St., Toronto, ON M4L 3T4, Canada, russo@ryerson.ca) and Emma Scholey (Psych., Univ. of Surrey, Guildford, United Kingdom)

The current study examined the effect of hearing loss and hearing aids in older adults on the experience of music emotion. Fifty-four participants aged 60 and above were recruited, including 18 with normal hearing (NH), 18 with hearing impairment (HI), and 18 with hearing impairment who use hearing aids (HA). Arousal ratings and skin conductance responses were obtained from participants across 24 extracts of film music that were previously validated as conveying one of four emotions: happy, sad, fearful, and tender. These four emotions represent a crossing of arousal (high and low) and valence (positive and negative). For each group, an "arousal range" was calculated as the average arousal ratings of happy and fearful excerpts, minus the average arousal ratings of tender and sad excerpts. A similar scheme was used for each group to determine a "skin conductance range". For negatively-valenced music, groups did not differ with regard to arousal of skin conductance. For positively valenced music, NH and HA yielded a larger arousal range than did HI. A similar pattern emerged from the analysis of skin conductance range. Results will be discussed with regard to the acoustic cues to emotion in music and signal processing methods in hearing aids.

8:45

3aSC3. Prosody as a window into speaker attitudes and interpersonal stance. Marc D. Pell, Nikos Vergis, Jonathan Caballero, Maël Mauchand, and Xiaoming Jiang (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8th Fl., Montreal, QC H3A1G1, Canada, marc.pell@mcgill.ca)

New research is exploring ways that prosody fulfils different social-pragmatic functions in spoken language by revealing the mental or affective state of the speaker, thereby contributing to an understanding of speaker's meaning. Prosody is often pivotal in signaling speaker attitudes or stance in the interpersonal context of the speaker-hearer; in contrast to the vocal communication of emotions, the significance of prosodic cues serving an emotive or interpersonal function is often dependent on the type of speech act being made and other contextual-situational parameters. Here, we discuss recent acoustic-perceptual studies in our lab demonstrating how prosody marks the interpersonal stance of a speaker and how this information is used by listeners to uncover social intentions that may be non-literal or covert. Focusing on how prosody operates in the communication of politeness, ironic attitudes, and sincerity, our data add to research on the acoustic-perceptual characteristics of prosody in these interpersonal contexts. Future implications for how acoustic cues underlying "prosodic attitudes" affect the interpretive process during on-line speech processing and how they influence behavioral outcomes are considered.

9:05

3aSC4. Acoustics and emotion in tonal and non-tonal languages: Findings from individuals with typical hearing and with cochlear implants. William F. Katz, Cecilia L. Pak (Callier Ctr. for Commun. Sci. and Disord., The Univ. of Texas at Dallas, 1966 Inwood Rd., TX 75235, wkatz@utdallas.edu), and Sujin Shin (Callier Ctr. for Commun. Sci. and Disord., The Univ. of Texas at Dallas, Richardson, TX)

Talkers of tonal language, such as Mandarin, use the acoustic cues of fundamental frequency (F_0), amplitude, and duration to indicate lexical meaning as well as to express linguistic and emotional prosody. It has therefore been hypothesized that tonal language talkers have less prosodic "space" to signal emotional prosody using F_0, compared to non-tonal language talkers, and that these differences in prosodic processing should be evident in speech perception and production tasks. In addition, for talkers of both tonal and non-tonal languages, speaking rate interacts with emotional expression, with "sad" mood generally expressed with slower speech, while "happy" and "angry" moods are marked with faster speech rates. Despite the overall importance of speaking rate in signaling particular emotional moods, few data exist for speakers of tonal languages, such as Mandarin, in which lexical tones are typically specified for length. These issues can be addressed by analyzing and modelling data for individuals with cochlear implants, electronic hearing systems that provide relatively good temporal resolution but poor spectral resolution. Findings from our laboratory will be used to address models of prosody perception and production.

9:25

3aSC5. How do babies laugh? Disa Sauter (Dept. of Psych., Univ. of Amsterdam, Niewe Achtergracht 129B, Amsterdam 1018 XA, Netherlands, d.a.sauter@uva.nl), Bronwen Evans (Univ. College London, London, United Kingdom), Dianne Venneker, and Mariska Kret (Leiden Univ., Leiden, Netherlands)

Laughter occurs across all great ape species, yet human laughter differs from that of other primates: Human laughter is primarily produced on the exhale, whereas other primates laugh on both the inhale and exhale. In the current study, we asked whether human infants laugh in a similar manner to apes, given that human infants, like non-human primates, tend to laugh in the context of tickling or rough-and-tumble play. Human adults, in contrast, laugh across many different kinds of social interactions. To test this hypothesis, we examined whether human infant laughter is acoustically more similar to non-human apes' laughter. Laughter clips from infants aged 3 to 18 months were annotated by phoneticians and evaluated by two listener samples (naïve listeners and phoneticians, respectively). The results provide support for the prediction that the proportion of infants' laughter produced on the exhale increases with age. These results suggest that at younger ages, human infants' laughter is more similar to that of other great apes. These findings are discussed in the context of vocal control maturation and social learning.

9:45–10:00 Break

10:00

3aSC6. Perception and production of vocal emotions by listeners with normal hearing and with cochlear implants. Monita Chatterjee, Jenni Sis, Sara Damm, and Aditya M. Kulkarni (Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, monita.chatterjee@boystown.org)

Informative cues for vocal emotions in speech include voice characteristics, speaking rate and intensity. For listeners with cochlear implants, voice characteristics such as pitch and vocal tract length are weakly represented in the electric signal; intensity cues are weakly to moderately preserved, and speaking rate is well represented. Our studies indicate that school-age children (6–18 years of age) with cochlear implants have significant deficits in their recognition of vocal emotions relative to normally hearing peers, similar to post-lingually deaf adults with cochlear implants. However, in contrast to the post-lingually deaf adults, children with cochlear implants showed large deficits in their *productions* of vocal emotions (happy/sad contrasts), measured in 1) acoustic analyses of the produced speech and 2) normally hearing listeners' perceptions of the produced emotions. In contrast to emotions, the words and sentences produced by the children with cochlear implants were highly intelligible. Post-lingually deaf adults with cochlear implants, however, showed excellent performance in the voice emotion production task. These results suggest that the perception of vocal emotions by listeners with cochlear implants may be limited primarily by device limitations, while emotion productions in speech are shaped more strongly by early auditory experience.

10:20

3aSC7. Acoustic cues for vocal emotion recognition by normal-hearing listeners and cochlear implant users. Xin Luo and Kathryn Pulling (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave., Tempe, AZ 85287, xinluo@asu.edu)

Our recent study found that cochlear implant (CI) users' quality of life in auditory, psychological, and social functioning were predicted by vocal emotion rather than sentence recognition scores. To eventually improve vocal emotion recognition with CIs, this study investigated the acoustic cues for vocal emotion recognition by CI users with CI alone or bimodal fitting, as compared to normal-hearing (NH) listeners. Sentence duration, mean fundamental frequency (F0), and F0 range were individually normalized for emotional utterances of each talker and sentence. In two other conditions, emotional utterances were presented backward in time and with upside-down F0 contours, respectively. Perceptual results showed significant effects of subject group, cue condition, talker, and emotion. Time-reversed utterances worsened NH listeners' recognition of all emotions except sad, while upside-down F0 contours worsened that of angry and happy. Vocal emotion recognition with CI alone only degraded with time-reversed utterances. Time-reversed utterances worsened bimodal CI users' recognition of angry and neutral, while upside-down F0 contours worsened that of angry and happy. Bimodal CI users and NH listeners were also affected by mean F0 and F0 range normalization when recognizing happy. We conclude that natural F0 contours should be faithfully encoded with CIs for better vocal emotion recognition.

10:40

3aSC8. Decoding musical and vocal emotions. Sebastien Paquette (Neurology, BIDMC - Harvard Med. School, 330 Brookline Ave., Palmer 127, Boston, MA 02215, spaquet1@bidmc.harvard.edu)

Many studies support the idea of common neural substrates for the perception of vocal and musical emotions. It is proposed that music, in order to make us perceive emotions, recruits the emotional circuits that evolved mainly for the processing of biologically important vocalizations (e.g., cries, screams). Although some studies have found great similarities between voice and music in terms of acoustic cues (emotional expression) and neural correlates (emotional processing), some studies reported differences specific to each medium. However, it is possible that the differences described may not be specific to the medium, but may instead be specific to the stimuli used (e.g., complexity, length). To understand how these vocal and musical emotions are perceived and how they can be affected by hearing impairments, we assessed recognition of the most basic forms of auditory emotion (musical/vocal bursts) through a series of studies in normal hearing individuals and in cochlear implant users. Multi-voxel pattern analyses of fMRI images provide evidence for a shared neural code for processing musical and vocal emotions. Correlational analyses of emotional ratings helped highlight the importance of timbral acoustic cues (brightness, energy, and roughness) common to voice and music for emotion perception in cochlear implant users.

11:00

3aSC9. Comparing theory, consensus, and perception to the acoustics of emotional speech. Peter M. Moriarty (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, moriarty@psu.edu), Michelle Vigeant (Graduate Program in Acoust., The Penn State Univ., State College, PA), Pan Liu (Univ. of Western ON, State College, PA), Rick Gilmore, and Pamela Cole (Dept. of Psych., The Penn State Univ., State College, PA)

A new corpus of emotional speech was created to conduct theory and evidence-based comparisons to the published literature on the effects of a speaker's intended emotion on the acoustics of their speech. Fifty-five adults were recorded speaking scripts with happy, angry, sad, and non-emotional prosodies. Variations in acoustic parameters such as pitch, timing, and formant deviations were investigated. Based on Scherer's (1986) theoretical predictions about differences between discrete emotions, and Juslin and Laukka's (2003) empirically-derived meta-analytic conclusions, we measured the degree to which the emotional speech data reflected predicted differences in the acoustic parameters of these prosodies. First, in relation to non-emotional prosody, angry and happy prosody were each 75% consistent with theory and sad prosody was not (25%). Second, in relation to non-emotional prosody, angry, happy, and sad prosodies were consistent with the empirical evidence base, 70%, 90%, and 50%, respectively. A subjective study was conducted wherein 30 adults rated the speech samples. Overall, adults discriminated the intended emotion of the speaker with 92% accuracy. Multiple regression analyses indicated that less than 25% of the significant acoustic patterns for each prosody accounted for variance in perceived emotional intensity. [Work supported by an award to Pamela Cole (NIMH 104547).]

11:20

3aSC10. Vocal emotion identification by older listeners with hearing loss. Huiwen Goy and Frank A. Russo (Psych., Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, huiwen.goy@psych.ryerson.ca)

A speaker's emotional state is one important type of information carried by the speech signal. Numerous studies have been conducted on younger and older normal-hearing listeners' ability to identify vocal emotions. Much less is known about how hearing loss affects emotion identification, and whether listeners with hearing loss use similar acoustic cues as normal-hearing listeners to identify emotions. The purpose of this study was to evaluate vocal emotion identification performance by younger normal-hearing listeners and older listeners who use hearing aids, and to examine possible acoustic cues used by listeners to distinguish between emotions. The results showed that older listeners with hearing loss performed much worse in a quiet listening environment (38.9%) than young normal-hearing listeners in quiet (84.7%) or in noise (68.4%), and that the use of hearing aids did not improve their performance significantly. Consideration of the pattern of identification errors and the acoustics of the stimuli suggested that the two listener groups relied on different F0-related cues to distinguish emotions: young listeners related on F0 mean and F0 contour, while older listeners relied mainly on F0 mean. Understanding how listeners with hearing loss identify emotion would provide guidance for developing rehabilitative strategies for hearing loss.

11:40

3aSC11. Processing of emotional sounds conveyed by the human voice and musical instruments. Sophie Nolden (Goethe-Univ. Frankfurt am Main, Theodor-W.-Adorno-Platz 6, Frankfurt am Main 60629, Germany, nolden@psych.uni-frankfurt.de), Simon Rigoulot (Int., Lab. for Brain, Music and Sound Res., Montreal, QC, Canada), Pierre Jolicoeur (Univ. of Montreal, Montreal, QC, Canada), and Jorge L. Armony (McGill Univ., Montreal, QC, Canada)

We investigated how emotional sounds are processed by musicians and non-musicians. Expertise through musical training can shape the way emotional music is processed. However, emotional sounds can arise from other sound sources, too, such as vocal expressions like laughter or cries. Musical expertise can also influence the way these emotional sounds are processed, thus, the impact of musical expertise can generalize over a wide range of emotional sounds. A recent study investigated oscillatory brain activity related to the processing of emotional sounds in musicians and non-musicians (Nolden, Rigoulot, Jolicoeur, & Armony, 2017, see also Rigoulot, Pell, & Armony, 2015). Musicians showed greater overall oscillatory brain activity than non-musicians. This was true for emotional music as well as for emotional sounds produced by the human voice, suggesting that expertise in processing one domain generalizes to different domains.

WEDNESDAY MORNING, 7 NOVEMBER 2018 SALON B (VCC), 8:30 A.M. TO 11:20 A.M.

Session 3aUWa

Underwater Acoustics, Acoustical Oceanography, Animal Bioacoustics, and Physical Acoustics: Biological Effects on Seabed Geoacoustic Properties

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

Megan S. Ballard, Cochair
Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78758

Kelly M. Dorgan, Cochair
Dauphin Island Sea Lab, Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528

Chair's Introduction—8:30

Invited Papers

8:35

3aUWa1. The trait-based approach as a tool to study the bio-geo seafloor system. Anna Törnroos (Environ. and Marine Biology, Åbo Akademi Univ., BioCity, Tykistönkatu 6, Turku 20520, Finland, anna.m.tornroos@abo.fi)

One of the major causes of heterogeneity on the seabed is the biology. The presence of organisms creates voids and frameworks within and on the sediment, and their behavior may layer or sort the entire seafloor. Making use of the biological information would be powerful for improving acoustics. Likewise, embracing, to a greater degree, acoustic techniques and measurements to understand the biology would be favorable. However, to tackle this bio-geo diversity in a cross-disciplinary way requires a common language and approach. Here I present and propose the trait-based approach as a way forward. Because there are simply too many species to describe and include in one model, reducing this complexity is essential and can be done by considering individuals characterized by a few key characteristics, or *traits*. Relevant biological traits span morphology, behavior and life-history of organisms and can be applied on single individuals and scaled up to whole communities, incorporating the density of organisms. By exemplifying the progress in benthic ecology, I outline where we currently stand, possible key traits of value for both fields and ideas on how to progress.

8:55

3aUWa2. Effects of burrow and tube construction by infauna on sound propagation through marine sediments. Kelly M. Dorgan, Will M. Ballentine, Grant Lockridge (Dauphin Island Sea Lab, Dauphin Island Sea Lab, 101 Bienville Blvd., Dauphin Island, AL 36528, kdorgan@disl.org), Megan S. Ballard, Kevin M. Lee (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs, Univ. of Texas at Austin, Austin, TX)

Infauna, animals living in marine sediments, modify sediment structure by burrowing, constructing burrows and tubes, and irrigating burrows. These activities can change the bulk porosity and density as well as create heterogeneity in sediment structure. We test the hypothesis that these activities alter sound speed and attenuation in sediments by manipulating homogenized sediments to mimic animal

activities. Specifically, we examine the effects of burrow excavation, burrow wall compaction, burrow irrigation, and construction of tubes from shell hash on sound speed and attenuation at 100, 200, and 400 kHz. Wavelengths corresponding to these frequencies span the size of the burrows constructed, and measurements were conducted at several depths within the upper 10 cm of sediment in which infauna are commonly found. Each of these activities or functions is performed by multiple species of animals that comprise a functional group. Our results will help identify functional groups that have important impacts on sediment acoustics and will be used to interpret field data in which deviations from predicted sound speed and attenuation are correlated with different and diverse communities of infauna.

Contributed Paper

9:15

3aUWa3. Effects of benthic biology on geoacoustic properties of marine sediments. Kevin M. Lee, Megan S. Ballard, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), Gabriel R. Venegas, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Infauna dwell in the benthic zone and have the capacity to modify the physical and acoustic properties of the seabed through bioturbation. To investigate such effects, *in situ* measurements of compressional and shear wave speed and attenuation were conducted in Petit Bois Pass, near the mouth of Mobile Bay, Alabama, USA. The measurement system was deployed at multiple locations within the pass, and acoustic measurements were conducted at depths up to 20 cm into the sediment to scan the portion of the seabed where most infauna live. Additionally, diver cores were collected and analyzed for infauna abundance and geotechnical properties, such as porosity and grain size distribution, for comparison with the acoustic data. While sediment geoacoustic models do not explicitly account for effects of biology, they do allow for parameterization of various physical properties like porosity, grain size, or pore fluid viscosity, all of which can be modified by the presence of biological organisms and bioturbation. Results from the *in situ* acoustic measurements and core analysis will be compared with such models to determine if any additional insight into the acoustic effects of infauna can be gained. [Work supported by ONR and ARL:UT IR&D.]

Invited Papers

9:30

3aUWa4. Effects of benthic biology on normal-incidence reflection coefficient measurements. Marcia J. Isakson and Lucas Shearer (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Normal-incidence reflection coefficient measurements have been shown to be sensitive to sediment type based on the structure and amplitude of the returned signal. In contrast to other sediment classification methods such as coring, these measurements can be done quickly and easily over a large area. This type of sediment classification scheme can also be integrated into surveys by single- and multi-beam echo sounders. However, benthic biology can substantially change the normal-incident acoustic return possibly confounding the sediment classification methods while simultaneously providing a method of assessing benthic eco-systems. In this study, a collection of normal-incidence measurements taken in a variety of underwater environments will be analyzed for the effects of benthic biology. The areas surveyed include the Chesapeake Bay, the Arctic, the Western Mediterranean, and Gulf Coast. Each of the datasets will be considered for the presence of benthic biology and its affect on geo-acoustic properties. [Work supported by ONR, Ocean Acoustics.]

9:50–10:05 Break

Contributed Paper

10:05

3aUWa5. Observations of the effects of benthic biology in measurements from the Acoustic Coring System collected in the New England Mud Patch. Megan S. Ballard, Kevin M. Lee, Andrew R. McNeese (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Jason D. Chaytor (U.S. Geological Survey, Woods Hole Coastal and Marine Sci. Ctr., Woods Hole, MA), Allen H. Reed (Naval Res. Lab., Stennis Space Ctr., MS), Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), Gabriel R. Venegas, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This paper presents *in situ* measurements of sediment sound speed and attenuation collected in the New England Mud Patch using the Acoustic Coring System (ACS) with comparisons to sound speed measurements from a multi-sensor core logger (MSCL). The ACS uses rod-mounted piezoelectric cylinders mounted below the penetrating tip of a gravity corer to obtain a continuous record of sound speed as a function of depth as the corer penetrates the seabed. The MSCL measurements of the recovered sediment cores were acquired in 2 cm increments. In both the ASC and MSCL measurements, an elevated sound speed is observed in the upper 25 cm of the seabed which is the portion of the seabed where most infauna live. The sediments collected in the cores were analyzed for density, porosity, bulk organic matter, carbonate content, mineral composition, and grain size. Additional surficial sediment samples recovered using a box core and a multi-core were sieved for infauna collection. The organisms were preserved, identified, and classified according to acoustically relevant traits. The acoustic measurements are interpreted in terms of the physical measurements of cores and the abundance of different types of organisms present in the seabed. [Work supported by ONR.]

10:20

3aUWa6. Geoacoustic properties of seagrass-bearing sediments. Gabriel R. Venegas (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu), Kevin M. Lee, Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Abdullah F. Rahman (Coastal Studies Lab., The Univ. of Texas Rio Grande Valley, Brownsville, TX)

Seagrasses are commonly referred to as "ecosystem engineers" due to their ability to modify their environments to create unique ecosystems believed to be the third-most valuable in the world. While engineering nursery habitats for small invertebrates, fish, and microalgae such as diatoms, they also aid in water filtration and are capable of storing 11% of the ocean's organic carbon each year. Seagrasses mainly grow in shallow salty waters within sandy sediments, but over time can drastically alter the sediment by entrapping silt particles, organic matter and decaying organisms that collect on the seafloor to form an organic-rich mud that incorporates itself with the underlying sand. Anaerobic microbes break down the buried organic matter to produce biogas and more stable organic compounds, which can drastically affect sediment geoacoustic properties. Sediment cores were collected within a *Thalassia testudinum* meadow in the Lower Laguna Madre, TX, and analyzed in 2-cm depth increments for grain size, density, porosity, sound speed, and attenuation from 100 kHz to 300 kHz, and organic carbon content. Results are compared with those from a seagrass-free sediment core to investigate how these valuable "ecosystem engineers" can alter geoacoustical properties of the seabed. [Work supported by ONR and ARL:UT IR&D.]

10:40

3aUWa7. High-frequency acoustic scattering from aquatic plants and seagrass leaf blades. Jay R. Johnson (Mech. Eng., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu), Kevin M. Lee, Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Jean-Pierre Hermand (LISA Environ. HydroAcoust. Lab, Université libre de Bruxelles (ULB), Brussels, Belgium), and Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Aquatic plants and seagrasses play a vital role in the littoral zone and shallow water ecosystems. They are important for carbon sequestration, habitat health, and seabed stabilization. Both aquatic plants and seagrasses are vascular plants with gas-filled internal channels, known as aerenchyma, with complex acoustic responses. Many previous field and laboratory experiments have shown that the acoustic response of vascular aquatic vegetation depends on tissue gas content, leaf blade structure, plant density, and photosynthetic activity, yet forward predictive models for acoustic propagation in aquatic plants and seagrass have not been realized. To help progress towards such a model, laboratory bistatic acoustic scattering measurements at 1 MHz were performed on aquarium raised *Vallisneria spiralis*. The measured results are compared to forward scattering measurements, at the same frequency, performed on naturally collected Mediterranean seagrasses *Posidonia oceanica* and *Cymodocea nodosa*. Microscopic images of leaf blade cross-sections were used to investigate the effect of tissue structure on acoustic response and nascent analytic and numerical models are presented to help interpret the measured results. [Work supported by ARL:UT IR&D, ONR, ONR Global, FNRS.]

11:00

3aUWa8. Effect of the microphytobenthos photosynthesis on seabed backscattering properties. Natalia Gorska, Adam Latala, and Filip Pniewski (Inst. of Oceanogr., Faculty of Oceanogr. and Geography, Univ. of Gdansk, al. Marszalka Pilsudskiego 46, Gdynia 81 - 378, Poland, oceng@ug.edu.pl)

The development of hydroacoustic techniques for benthic habitat classification requires understanding of biological effects on seabed geoacoustic properties. The study addresses only one biological process that could potentially change the properties—photosynthesis of benthic microalgae. The previous investigations demonstrated that the impact could be important in warm southern marine waters of relatively high salinity. This motivated us to study the effect of the microphytobenthos photosynthesis on the backscattering properties of the sandy sediments of the southern Baltic Sea, an area of lower temperature and salinity. The five multiday laboratory experiments, different in hydrophysical or biological conditions, were conducted. The backscatter data were acquired in the small tank with sandy bottom under controlled constant temperature and salinity with simulated "day" and "night" conditions (light/dark (L/D) photocycles). Oxygen content in the water column as well as biological and biooptical parameters were additionally monitored. The diel variations of the backscattered signal energy were analyzed. The study demonstrated the impact of microphytobenthos photosynthesis on the backscattering properties of the marine sediment and its sensitivity to the abiotic (illumination level) and biotic (benthic microalgal biomass and macrozoobenthos bioturbation) factors.

Session 3aUWb

Underwater Acoustics: Topics and Modelling in Underwater Acoustics

Darrell Jackson, Cochair
Applied Physics, University of Washington, 1013 NE 40th St., Seattle, WA 98105

Dale D. Ellis, Cochair
Physics, Mount Allison University, 18 Hugh Allen Drive, Dartmouth, NS B2W 2K8, Canada

Contributed Papers

9:00

3aUWb1. The mutual interface scattering cross section. Darrell Jackson and Dajun Tang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, drj@apl.washington.edu)

Scattering by the ocean surface or seafloor is often characterized by the interface scattering strength, the decibel equivalent of the scattering cross section per unit solid angle per unit area (referred to subsequently as the interface scattering cross section for brevity). The scattering cross section is a statistical second moment that allows modeling of the mean-square scattered pressure. Use of the interface scattering cross section requires that the incident field can be approximated by a single plane wave, a condition that is not satisfied when the interface is ensonified in the near field of a large array, as in synthetic-aperture sonar (SAS), or when multipath propagation is important as in modeling reverberation. The mutual cross section introduced in this presentation is a generalization of the interface scattering cross section to include ensonification by multiple plane waves. Examples of application to SAS and the ocean waveguide will be shown. In the latter case, it is seen that the mutual cross section can model peaks in the reverberation time series that occur as a result of multipath convergence on the interface. (Work supported by ONR.)

9:15

3aUWb2. Autoproducts and out-of-band acoustic fields in refracting multipath environments with caustics. Brian M. Worthmann (Appl. Phys., Univ. of Michigan, 1231 Beal Ave., 2010 Automotive Lab, Ann Arbor, MI 48104, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Autoproducts are nonlinear mathematical constructions developed from a given acoustic field with non-zero bandwidth that can—with some limitations—mimic lower and higher frequency acoustic fields lying outside the bandwidth of the given acoustic field. The autoproducts' mimicry of out-of-band fields is often sufficient for successful beamforming and/or source localization at below- or above-band frequencies chosen to correct array sparseness, suppress random scattering, mitigate environmental mismatch, and/or enhance resolution. While the limitations of autoproducts developed from acoustic fields well-described by ray acoustics are understood, what is less studied are the effects of diffraction and refraction. In this presentation, acoustic waveguide environments are defined which contain strong refraction, and form caustics—regions where nearby ray paths cross—which results in a phase shift in the propagating waves. These caustic phase shifts are not found in autoproduct fields and are detrimental for autoproduct-based remote sensing, especially in regions that are reached by multiple rays that have passed through different numbers of caustics. Multiple refracting environments are considered, including n^2-linear and n^2-quadratic sound speed profiles, and the ability of autoproducts to mimic out-of-band fields in these environments is assessed. Implications for autoproduct-based source localization are discussed. [Sponsored by ONR and NSF.]

9:30

3aUWb3. An improved Gaussian beam caustic correction for Bellhop at low frequencies. Diana F. McCammon (McCammon Acoust. Consulting, Waterville, NS, Canada) and Dale D. Ellis (Phys., Mount Allison Univ., 18 Hugh Allen Dr., Dartmouth, NS B2W 2K8, Canada, daledellis@gmail.com)

Gaussian beams are commonly used in ray-tracing to mitigate the effects of caustics and shadow zones. The problem is how to determine the width of these "fuzzy" beams. Porter and Bucker [J. Acoust. Soc. Am. 82, 1349–1359, 1987] proposed a method that expressed the beamwidth and curvature in terms of p and q of the dynamic ray equations. We call q the beamwidth factor. In the Gaussian beam implementation caustics are not caused by the crossing of two rays; rather they occur when the beamwidth factor, which appears in the denominator of the amplitude, becomes small. In practice, this is generally not a problem at high frequencies, but as the frequency gets lower the problem gets more severe. The widely used Bellhop model has a procedure, which "caps" the beamwidth when q becomes too small, but the procedure eventually breaks down at low frequencies. Here, we propose a different cap based on a cylindrical wave front converging to the focal point of a caustic. The various caps are compared with the "exact" normal mode solution for a shallow-water upward refracting environment, illustrating how the new cap provides better reduction of the caustic anomalies.

9:45

3aUWb4. Modeling of the power delay profile for underwater acoustic channels—Effects of out-of-plane scattering and reverberation. Trond Jenserud (Norwegian Defence Res. Establishment (FFI), Karljohansvern, Horten NO-3191, Norway, trond.jenserud@ffi.no) and Sven Ivansson (Swedish Defence Res. Agency (FOI), Kista, Sweden)

Long reverberation tails seem to be a common feature of shallow water acoustic channel impulse responses. We show examples from two significantly different environments along the Norwegian coast, and demonstrate that 2-D propagation modeling fails to reproduce the long tails observed. The continuous nature of the impulse responses and the low correlation and slow decay rates of the tails point towards reverberation as an important mechanism behind the observations. The effects of reverberation and out-of-plane scattering are then included in the modeling by employing the 3-D ray-based model Rev3D. Scattering from the sea surface and sea bottom are described by scattering-strength functions including azimuthal variation. The modeling results indicate that the long reverberation tails can be explained by 3-D out-of-plane scattering. Although the measured levels are not perfectly matched, the importance of including reverberation in the modeling is clearly demonstrated.

10:00–10:15 Break

10:15

3aUWb5. Model and data comparison for a three-dimensional, finite-difference, time-domain solver in Sequim Bay. Erin C. Hafla (Civil Eng., Montana State Univ., 205 Cobleigh Hall, Bozeman, MT 59717-3900, erinhafla@gmail.com), Erick Johnson (Mech. Eng., Montana State Univ., Bozeman, MT), Jesse Roberts (Sandia National Lab., Albuquerque, NM), and Kaustubha Raghukumar (Integral Consulting, Inc., Santa Cruz, CA)

Paracousti is a three-dimensional finite-difference, time-domain solution to the governing velocity-pressure equations. This program is directed at modeling sound propagation generated by marine hydrokinetic (MHK) sources in an ocean environment. It is capable of modeling complex, multi-frequency sources propagating through water and soil that have spatially varying sound speeds, bathymetry, and bed composition. Experimental sound data collected at Sequim Bay in Washington, USA, during the winter of 2017 is compared against several simulations modeled within Paracousti for a range of frequencies and receiver locations. This measurement campaign recorded ambient noise data and the sound from a source producing three-second long, sinusoidal pulses between 20 and 5,000 Hz at a depth of 3 m. Additionally, bathymetric, sanity, and temperature data for the bay were collected in order to calculate the sound speed. Data were recorded at six locations ranging in distance between 10 and 1,000 m from the source by stationary buoys. Each simulation was created to model the collected source profiles and has a total depth of 80 m, with the average soil depth occurring at 23 m, and compared via transmission losses.

10:30

3aUWb6. Investigation of property losses in single crystal transduction devices. Michael Warnock and Thomas R. Howarth (Naval Undersea Warfare Ctr., 25 Merton Rd., Newport, RI 02840, michaelwarnock4@gmail.com)

With the advent of relaxor single crystal materials, such as lead indium niobate-lead magnesium niobate-lead titanate (PIN-PMN-PT) and lead magnesium niobate-lead titanate (PMN-PT), it was discovered that the highly oriented crystal structure of the materials leads to large values of dielectric and piezoelectric properties favorable to acoustic applications. When utilized in transduction devices, single crystal materials have demonstrated better acoustic performances compared to traditional designs. However, during construction of single crystal transducers, a capacitance loss is measured after mechanically bonding the crystal to a stiff backing material. The lower capacitance suggests a decrease in the effective permittivity, decreasing favorable piezoelectric constants for acoustic performance. In this study, the loss in effective permittivity was modeled using a commercial finite element analysis software. We present simulations used to demonstrate the loss in effective permittivity and compare with experimental measurements. Additionally, finite element simulations of several design considerations that can be utilized by transducer designers to minimize the loss in the effective permittivity are presented.

10:45

3aUWb7. Soundscape characteristics in Southern Resident Killer Whale critical habitats. Svein Vagle (Fisheries and Oceans Canada, 9850 West Saanich Rd., Sidney, BC V8L4B2, Canada, Svein.Vagle@dfo-mpo.gc.ca), Caitlin O'Neill (Fisheries and Oceans Canada, Victoria, BC, Canada), Sheila Thornton, and Harald Yurk (Fisheries and Ocean Canada, West Vancouver, BC, Canada)

The Southern Resident Killer whales (Orcinus orca) (SRKW) are an endangered group of orcas with current range of Pacific North East from California to Northern British Columbia and spend most of the summer months in and around the Salish Sea. This group of mammals feed primarily on fish, are very local, and live in tight-knit family units called pods. July 1 2017 census reported 77 animals; which now has been reduced to 76 by a more recent death. Anthropogenic underwater noise, primarily from commercial and recreational vessels is suspected to have detrimental effects on these whales. Here, we present results from a seven month, continuous sound recording (125 kHz acoustic bandwidth), whale centered study to monitor and interpret different soundscapes in SRKW critical habitats. The six different locations studied have significantly different acoustic transmission characteristics and cover open ocean areas, where natural sound spectral levels are high, to areas where anthropogenic noise-sources dominate. Possible implications of these different characteristics on the ability of the orcas to communicate and find prey are discussed. [Work funded by the Ocean Protection Plan (OPP) of the Government of Canada.]

11:00

3aUWb8. Grid-free compressive DOA estimation via alternating projection. Yongsung Park (Seoul National Univ., Gwanak-gu, Gwanak-ro 1, Seoul National University Bldg. 36 - Rm. 212, Seoul 08826, South Korea, ysparkwin@snu.ac.kr), Mark Wagner, Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), and Woojae Seong (Seoul National Univ., Seoul, South Korea)

On a discrete angular grid, the compressive sensing (CS) reconstruction degrades due to basis mismatch when the direction-of-arrivals(DOAs) do not coincide with the angular directions on the grid. To overcome this, we solve the DOA estimation in the grid-free angular domain. A method for solving the grid-free CS based beamforming problem is presented. The grid-free CS beamforming involves using a convex relaxation of a sparsity exploiting method and solving its semidefinite characterization. Current semi-definite programming solvers have high computational complexity for problems formed from typical large datasets. We solve the non-convex formulation of the grid-free CS beamforming optimization problem with alternating projections (AP). The AP based grid-free CS beamforming can solve large scale problems with hundreds of observations in a few seconds. We compare and contrast our algorithm with CS based beamforming as well as conventional beamforming.

11:15

3aUWb9. Validation of airgun array modelled source signatures. Craig McPherson (JASCO Appl. Sci., Unit 4, 61-63 Steel St., Capalaba, QLD 4157, Australia, craig.mcpherson@jasco.com), Alexander O. MacGillivray (JASCO Appl. Sci., Victoria, Br. Columbia, Canada), and Edward Hagar (Polarcus Asia Pacific, Singapore, Singapore)

Characterizing the source levels from seismic airgun arrays to increase confidence in model accuracy has been identified as a requirement by the scientific and regulatory community. To comply with regulatory requirements in Australia, a measurement program was conducted to validate the source signature predictions of JASCO's Airgun Array Source Model. The validation program measured a range of commercial airgun arrays, including a 2380in^3 array, and was conducted in 80m of water with array passing directly over the recorder on the seafloor. For the 2380in^3 array, the maximum measured PK was 220.62 dBre1μPa. Due to the proximity of the measurements, interesting features such as ghost cavitation and airgun recharge were observed in the data. The measurement study results were used to validate the modelled far-field source levels through a comparison between the measured received sound-levels and predicted received sound-levels at a real receiver point in the far-field of the source. The predictions were made using a wavenumber integral model coupled to the airgun source model. The program measured received sound levels in the endfire, broadside and vertical directions, and as the results showed good agreement with the modelling results, provided a validation of the complete modelled source signatures for the array.

Session 3pAA

Architectural Acoustics: Absorption, Diffusion, and Insulation

Shane J. Kanter, Cochair
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Chair's Introduction—1:00

Contributed Papers

1:05

3pAA1. Cross-stitched fabric as potential sound absorber for room acoustics applications. Xiaoning Tang and Xiong Yan (College of Textiles, Donghua Univ., Renmin North Rd. 2999, Shanghai 201620, China, tangxn123@163.com)

Textiles materials are widely used in noise and vibration control engineering. A thin fabric absorber consists of an air cavity and a backing rigid wall, and its main applications include ceiling, curtain, and sound barrier, etc. The main aim of this work is to study the acoustical and sound absorption properties of cross-stitched fabric, which has a similar structure with micro-perforated panel. The sound absorption coefficients of cross-stitched fabrics with different specifications were measured by impedance tube method, where the air cavity distance is 1 cm, 2 cm, and 3 cm, respectively. These distances are suitable to the binding and layout design of cross-stitched fabric painting in practical applications. As industrial textiles, cross-stitched fabric can be made into diversity of artistic patterns through manual or mechanized embroidery technology. This work has also studied the effects of multifarious embroidered patterns on the sound absorption coefficients of fabrics with different air cavity distance. In summary, cross-stitched fabric with exquisite patterns is expected to be a promising sound absorber for room acoustics.

1:20

3pAA2. Wave-based modeling of sound insulation of a double layer finite panel with different infilling materials between two reverberant rooms. Fangliang Chen, Yihe Huang, and Tejav DeGanyar (Virtual Construction Lab, Schuco-USA, 260 West 39th St., Ste. 1801, New York, NY 10018, fchen@schuco-usa.com)

A wave-based model is presented in this study to investigate the direct sound transmission through a double layer panel infilled with different materials and embedded between two rooms. The modal behavior of both the rooms and elastic finite layers are considered; accordingly, the full vibro-acoustic coupling between rooms, panels, and air cavity are accounted for. To validate the presented model, experiments on a double layer finite panel with two aluminum face sheets infilled with different absorption materials between two reverberant rooms are conducted at a certificated testing chamber. Comparisons between the results predicted by the wave-based model and those obtained by INSUL are presented as well. Comprehensive parametric studies are further conducted to study the effects of different design parameters on the sound transmission loss of a double layer panel., such as different filling materials, panel locations in the partition wall, cavity dimensions, boundary conditions of the panel embedded in the wall, damping factor of the elastic layers, panel dimension, and aspect ratio. Findings from the parametric studies will provide an important practice value to guide the design and assembling of a desired double layer panel to achieve an optimized performance of sound insulation between two adjacent rooms.

1:35

3pAA3. Sound absorption through PDMS-based flexible perforated plate backed by a dual layer cavity. Sanjay Kumar, Pulak Bhushan, and Shantanu Bhattacharya (Dept. of Mech. Eng., Indian Inst. of Technol. Kanpur, Microsystems Fabrication Lab., Kanpur, Uttar Pradesh 208016, India, sanjay21505@gmail.com)

A structure consisting of a perforated plate backed by an air cavity is a widely used structure for sound absorption in the low-to-high frequency range. In this study, we propose a sound absorbing metastructure composed of a polydimethylsiloxane (PDMS)-based flexible perforated plate backed by a layer of multiple hexagonal unit cells. Each unit cell consists of a dual cavity connected by a rigid perforated plate. The proposed structure allows for enhanced vibroacoustic coupling between the perforated plate and the back cavity. First, the structure is theoretically simulated using the finite element based analysis. The results elicit that the shape and size of the back cavity significantly alter the sound absorption coefficient and the resonant frequency of the structure. The geometrical parameters of the metastructure were further optimized using an artificial intelligence-based technique. Using the optimized geometrical parameters, the back cavity was printed using selective laser sintering (SLS) and a PDMS perforated plate has adhered on top of it. The experimental results demonstrate a potential sound absorption metastructure which can be utilized for multiple applications.

1:50

3pAA4. A diffuser language: Designing quadratic residue diffuser arrays with shape grammars. Jonathan Dessi-Olive and Timothy Hsu (Architecture, Georgia Inst. of Technol., School of Architecture - Georgia Tech, 245 4th St., NW, Ste. 351, Atlanta, GA 30332, jdo@design.gatech.edu)

This paper presents research on a rule-based approach to designing creative acoustic diffuser arrays. A shape grammar-influenced design method is specified that uses shape rules to recursively design arrays of quadratic residue diffusers (QRD) in a way that is neither mechanical nor deterministic. Shape grammar-based generative systems have already been shown to be capable of being used in architecture and engineering to create languages of

3p WED. PM

functional objects but have not yet been extended into the realm of acoustics. The grammar presented by this paper produces a QRD-based acoustic diffusion language that breaks habits in QRD deploying techniques, which is lacking outside of using known equations that give known forms. The grammar can include different design frequencies and open the possibilities of different and non-uniform, intentional diffusion treatment. This paper will specify and demonstrate the formation rules, and discuss on how in combination with material specifications, this design method can generate broadband QRD arrays that address both performative and aesthetic criteria in new and surprising ways.

2:05

3pAA5. Acoustic impedance control for reproducing sound environment by constructing virtual wall. Eiji Yokota and Seiichiro Katsura (Integrated Design Eng., Keio Univ., 3-14-1, Hiyoshi, Kohoku, Yokohama, Japan, Yokohama, Kanagawa 223-8522, Japan, yokota@katsura.sd.keio.ac.jp)

In order to obtain high presence, it is necessary to reproduce the sound environment of remote places. Conventionally, sound environments at remote places are reproduced by recreating a sound signal at that place. In this study, an acoustic technique that reproduces sound environments of remote places by recreating the dynamics of the environment at that place is proposed. In the proposed acoustic technique, the desired sound environment is attained by making listeners on the spot feel as if there were a wall that reflects and absorbs the sound waves. By controlling the acoustic impedance at the position where the wall is constructed, the wall is virtually built. In addition, the acoustic impedance including the virtual viscosity and elasticity that can be arbitrarily determined at the position where the wall constructed is set. As a result, by deciding the acoustic output from the secondary sound source so as to attain the desired acoustic impedance, the wall is virtually constructed to reproduce the desired sound environment.

2:20

3pAA6. Comparative study of the sound absorption coefficient of two absorbent materials with two empirical methods for low frequency. Nelson L. Legat, Claudia C. Silva, Nilson Barbieri, and Key F. Lima (PUCPR, Imaculada Conceição, 1155, Curitiba, Paraná 80215901, Brazil, keyflima@gmail.com)

The equations of Delany and Bazley have been widely used in the study of the behavior of sound waves inside porous materials since 1970. However, as is known, the validity of these equations for low frequencies can produce physically inconsistent results. This work intends to evaluate the sound absorption coefficient of two types of absorbent material, one porous and one fibrous, with the Allard and Champoux's equations presented in 1992. These equations have the purpose of improving acoustic properties characterization at a low frequency. The precision of the results obtained through the two empirical methods is verified with an experimental evaluation using the modified two cavity method. The measurements are made on the impedance tube with the samples arranged in two different positions displaced from the rigid termination. Porous samples evaluated by this work were common foam and the fibrous were coconut coir fibers.

2:35

3pAA7. The acoustics of the "Witches Valley." Umberto Berardi (DAS, Ryerson Univ., 350 Victoria St., Toronto, ON M5B 2K3, Canada, uberardi@ryerson.ca), Gino Iannace, and Amelia Trematerra (Seconda Universita' di Napoli, Aversa, Caserta, Italy)

According to legend, witches were women with magical knowledge that with the use of herbs could cure illnesses or perform spells. In 1639, the book "De Nuce Maga Beneventana" was published and described how the place where the "witches" gathered was an area south in South Italy in a long, narrow gorge with high rock walls, called the "Stretto di Barba." The legend of the witches originated with the invasion of southern Italy by the Longbards who, according to legend, under a walnut tree performed sacred rituals with dances and chants accompanied by the rhythm of drums. To make the rituals more effective, mysterious, and emotionally involving for the people take, it was necessary to amplify the sounds generated. This could only take place in the presence of a narrow gorge with flat and parallel rock walls where the reflecting of the sound generated echoes. The aim of this paper is to verify whether the "Stretto di Barba" has the acoustic characteristics that create echoes and sound amplification effects to confirm the legend that the Longbards performed their rituals here.

WEDNESDAY AFTERNOON, 7 NOVEMBER 2018 SHAUGHNESSY (FE), 1:00 P.M. TO 3:30 P.M.

Session 3pAB

Animal Bioacoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Passive Acoustic Density Estimation: Recent Advances and Outcomes for Terrestrial and Marine Species II

Thomas F. Norris, Cochair
Bio-Waves Inc., 517 Cornish Dr., Encinitas, CA 92024

Tiago A. Marques, Cochair
School of Mathematics and Statistics, University of St. Andrews, The Observatory, Buchannan Gardens, St. Andrews KY16 9 LZ, United Kingdom

Chair's Introduction—1:00

Invited Papers

1:05

3pAB1. Sound localization for estimating population densities of birds: Progress, challenges, and opportunities. Richard W. Hedley and Erin M. Bayne (Biological Sci., Univ. of Alberta, 11335 SK Dr. NW, Edmonton, AB T6G 2M9, Canada, rhedley@ualberta.ca)

Autonomous recording units (ARUs) are seeing rapid adoption as a research tool to estimate population densities of birds. One challenge for estimating population density is that the distance to the vocalizing bird is unknown. An emphasis has been placed on assessing the detection radius for different species on ARUs, and more recent work has had success relating sound amplitude to distance. We argue, however, that the true variable of interest is not density, per se, but the relationship between density and habitat type or other variables. In an ideal world, ARUs would collect information on the absolute location of each sound source, rather than the number of each species within earshot of the microphone, since the detection radius of an ARU commonly encompasses multiple habitat types. Sound localization is a promising solution to this problem, with some localization methods allowing sound sources to be pinpointed within 1m of their true location. We will present localization results comprising several thousand songs from several species in the Boreal forest, to highlight how localization data can be translated into estimates of population density. We will also highlight several challenges and opportunities for future contributions in the field of sound localization.

1:30

3pAB2. Detection probability of Cuvier's beaked whale clicks from a glider and a deep-water float. Selene Fregosi (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Hatfield Marine Sci. Ctr., 2030 SE Marine Sci. Dr., Newport, OR 97365, selene.fregosi@oregonstate.edu), Danielle Harris (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR), Jay Barlow (Marine Mammal and Turtle Div., Southwest Fisheries Sci. Ctr., NOAA, La Jolla, CA), Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St Andrews, United Kingdom), and Holger Klinck (BioAcoust. Res. Program, Cornell Lab of Ornithology, Cornell Univ., Ithica, NY)

Work is being conducted to estimate marine mammal density and abundance from slow-moving, passive acoustically equipped underwater gliders and deep-water floats. We deployed five drifting acoustic spar buoy recorders (DASBRs) simultaneously with a seaglider and QUEphone float in the Catalina Basin of Southern California in 2016 to estimate the probability of detecting Cuvier's beaked whale echolocation clicks from the glider and float. The DASBRs successfully localized and tracked individual whales, though a limited number of encounters prohibited estimation of detection probability through a trial-based method. We explored using a spatially explicit capture recapture (SECR) approach. The SECR analysis was modified to account for the non-static array and we explored sample size requirements for SECR using simulations. During tracked dives, over 200 one-minute time bins contained echolocation clicks on at least one DASBR. Of these, 10 and 34 bins contained clicks recorded on the glider and float, respectively, and were used to make preliminary estimates of detection probability as a function of range. Detection probability estimation is a key step towards animal density estimation; other information such as encounter rate and click production rates are required to estimate animal density, which will form the next part of this work.

1:55

3pAB3. Estimating densities of terrestrial wildlife using passive acoustic recordings: A pragmatic approach using paired human observations. Steven Van Wilgenburg (Environment and Climate Change Canada, Canadian Wildlife Service, 115 Perimeter Rd., Prairie and Northern Wildlife Res. Ctr., Saskatoon, SK S7N0X4, Canada, steven.vanwilgenburg@Canada.ca), Péter Sólymos (AB Biodiversity Monitoring Inst., Edmonton, AB, Canada), Kevin Kardynal (Environment and Climate Change Canada, Sci. & Technol. Branch, Saskatoon, SK, Canada), and Matthew D. Frey (Univ. of SK, Saskatoon, SK, Canada)

The use of passive using acoustic monitoring is a rapidly growing field in terrestrial ecology, particularly in ornithology. Much recent work has focused on using recording technologies for large-scale occupancy monitoring, while local scale studies have focused on acoustic localization for density estimation. Density estimation offers numerous advantages for answering key conservation questions but may seem impractical to employ at national or continental scales using current acoustic localization techniques. Here, we describe how paired sampling can be used in conjunction with generalized linear (GLM) or generalized linear mixed models (GLMM) to estimate correction factors (δ) to derive density estimates from single acoustic recorders, provided that paired human observers conduct distance estimation. Thus, our approach provides an alternative to more complicated and expensive methods. We discuss the advantages and disadvantages of our approach and highlight the context in which we see various existing or emerging acoustic density estimation methods being used in the terrestrial environment, given the nature and scale of the ecological question(s) being asked.

2:20

3pAB4. An evaluation of density estimation methods using a multiyear dataset of localized bowhead whale calls. Katherine H. Kim (Greeneridge Sci., Inc., 90 Arnold Pl, Ste D, Santa Barbara, CA 93117, khkim@greeneridge.com), Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), Tiago A. Marques (Centro de Estatística e Aplicações, Faculdade de Ciências da Universidade de Lisboa, St. Andrews, Fife, United Kingdom), Danielle Harris, Cornelia S. Oedekoven (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), Gisela Cheoo (Departamento de Estatística e Investigação Operacional, Faculdade de Ciências da Universidade de Lisboa, Lisboa, Portugal), Aaron Thode (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Susanna B. Blackwell (Greeneridge Sci., Inc., Aptos, CA), and Alexander Conrad (Greeneridge Sci., Inc., Santa Barbara, CA)

For eight years (2007–2014), Greeneridge Sciences deployed, collected, and analyzed an immense passive acoustic dataset to study the response of bowhead whales (*Balaena mysticetus*) to oil exploration activities along a 280 km swath of the Alaskan Arctic continental shelf. Each year, up to 40 Directional Autonomous Seafloor Acoustic Recorders (DASARs) detected bowhead calls during the whales' annual fall migration, resulting in over 2.4 million localized calls over the study period. These enormous sample sizes, combined with the unique localization capability of the DASARs, have enabled the investigation and cross-validation of several acoustic density estimation (DE) methods. Here, we compared results among three density estimation methods: (i) direct census, (ii) distance sampling, and (iii) spatially explicit capture recapture (SECR). In the course of our investigation, we encountered and addressed two challenges: (i) how to apply point transect theory to distributed array systems, and (ii) how to implement SECR in situations where detections between sensors are not statistically independent or where false positives exist on single sensors. These density estimates were then used to estimate relative abundance of the Western Arctic bowhead whale population over multiple years. [Work sponsored by the Joint Industry Programme.]

Contributed Papers

2:45

3pAB5. Density estimation using ocean-bottom seismometers in Cascadia. David K. Mellinger (Coop. Inst.for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu)

The Cascadia Initiative placed a large number of ocean-bottom seismic sensors off the coast of the US Pacific Northwest. These sensors recorded pulsed sounds from fin whales, recorded in conditions that in many cases make it impossible to resolve the sounds from individual whales. The sounds from multiple whales, however, combine to make a distinct rise in spectral energy in the 15–30 Hz band. This received energy can be used to estimate the population density of fin whales in the vicinity of each sensor over the 4-year duration of the project. Difficulties addressed include the variable rate and intensity of fin whale vocalizations, acoustic properties of the seafloor, and differences between seismic and acoustic sensors. [Funding from ONR.]

3:00

3pAB6. Can whistles be used to estimate dolphin abundance? Carmen Bazúa Durán (Física, UNAM, Facultad de Ciencias, Circuito Exterior s/n, Ciudad Universitaria, México, D.F. 04510, Mexico, bazua@unam.mx)

Dolphin whistles have been studied extensively for both wild and captive animals. They are used in the communication between individuals, to maintain contact within individuals of a herd, and to coordinate herd movements. However, little is still known on how whistles can be used to ascribe individuals. Therefore, the present study is focused on determining the stereotypy of the most frequently emitted whistles by captive bottlenose dolphins,*Tursiops truncatus*, housed in two different marine parks. Stereotypy was computed by changing the similarity index while classifying whistles into whistle types using Matlab BELUGA and ArtWARP. Signature whistles are an individual distinction and have a similarity index greater than 95%, therefore, allowing to assess the minimum number of whistling dolphins in a pod that emit signature whistles. This is specially important in the wild, where in some occasions, it is very difficult to assess how many individuals are present. Thus this very simple method will be useful to quantify the number of signature whistles from underwater recordings, and to relate it with the possible number of dolphins present. It is necessary to implement methods like this one to better understand how dolphins are using whistles, since acoustic communication is the most important sense in dolphin species. [Work supported by PAPIIT&PASPA-UNAM.]

3:15

3pAB7. Acoustic versus visual monitoring of Cetaceans in coastal habitats. Benjamin T. Hendricks (Elec. and Comput. Eng., Univ. of Victoria, 3800 Finnerty Rd., Eng. Office Wing, Rm. 448, Victoria, BC V8P 5C2, Canada, hendrick@uvic.ca), Eric M. Keen (Marine Ecology and Telemetry Res., Seabeck, WA), Janie L. Wray (North Coast Cetacean Society , Hartley Bay, BC, Canada), Hussein M. Alidina (Oceans Program, World Wildlife Fund - Canada, Victoria, BC, Canada), Chris R. Picard (Gitga'at Oceans and Lands Dept., Hartley Bay, BC, Canada), Hermann Meuter (none, Hartley Bay, BC, Canada), and T. Aaron Gulliver (Elec. and Comput. Eng., Univ. of Victoria, Victoria, BC, Canada)

Monitoring the aquatic soundscape provides critical information about marine mammal habitat use. It is essential for developing mitigation strategies for areas with expanding marine shipping and industrial activity. A long baseline hydrophone array has been installed in Squally Channel, a culturally, ecologically, and economically important marine environment in northern British Columbia, Canada. The array consists of four synchronized bottom-mounted hydrophones that permanently record and radio-transmit data to a land-based laboratory in real-time. The acoustic monitoring is supplemented with a land-based visual observatory that oversees the same area of approximately 200 km^2. Automated detectors have been developed for vocalizations of humpback whales, orcas, and fin whales. Acoustic data and visual surveys were analyzed for a period of more than 100 days between 2017 and 2018. We present an overview of the acoustic detection functions and their performance by call type and summarize visual survey procedures. Cetacean activity derived from automated acoustic detections and visual observations is analyzed. Finally, acoustic and visual survey methods are compared to assess a) the fraction to which different species are acoustically active while in the area, and b) the effectiveness of both acoustic and visual monitoring efforts for the purpose of cetacean monitoring.

Session 3pBAa

Biomedical Acoustics and Physical Acoustics: Bubble Trouble in Therapeutic Ultrasound II

Christy K. Holland, Cochair
Internal Medicine, Division of Cardiovascular Diseases and Biomedical Engineering Program, University of Cincinnati, Cardiovascular Center Room 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586

Klazina Kooiman, Cochair
Thoraxcenter, Dept. of Biomedical Engineering, Erasmus MC, P.O. Box 2040, Room Ee2302, Rotterdam 3000 CA, Netherlands

Invited Paper

1:00

3pBAa1. Advances in controlled transcranial bubble-mediated drug delivery and opportunities for transvertebral therapy. Meaghan A. O'Reilly, Ryan M. Jones, Lulu Deng, Kogee Leung, Stecia-Marie P. Fletcher, Rui Xu, and Kullervo Hynynen (Sunnybrook Res. Inst., 2075 Bayview Ave., Rm. C713, Toronto, ON M4N3M5, Canada, moreilly@sri.utoronto.ca)

The treatment of central nervous system (CNS) disorders is hindered by the presence of specialized barriers that maintain the privileged CNS environment and in doing so restrict the transport of molecules from the vascular compartment to the parenchyma. The ability of ultrasound in combination with intravenously administered microbubbles to transiently open the Blood-Brain Barrier (BBB) to permit the delivery of therapeutic agents is well established in preclinical models and has reached the stage of clinical investigations. Although the methods are not yet as developed as those for BBB opening, ultrasound can also modify the Blood-Spinal Cord Barrier (BSCB) to facilitate drug delivery. Both treatments have immense potential to revolutionize the treatment of CNS disorders, but their widespread clinical adoption hinges on the establishment of robust methods to deliver, monitor and control the exposures to ensure safe and effective treatments in these sensitive tissues. This talk will present recent advances in methods for brain therapy, including three-dimensional transcranial microbubble mapping for treatment control, and will also present new preclinical findings in BSCB opening and approaches for controlled transvertebral focusing at clinical scale.

Contributed Papers

1:20

3pBAa2. Lipid-shelled microbubbles for ultrasound-triggered release of xenon. Himanshu Shekhar, Arunkumar Palaniappan (Dept. of Internal Medicine, Univ. of Cincinnati, 3933 Cardiovascular Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267, himanshu.shekhar@uc.edu), Tao Peng, Melanie R. Moody, Shaoling Huang (Dept. of Internal Medicine, Univ. of Texas Health Sci. Ctr. at Houston, Houston, TX), Kevin J. Haworth (Depart. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), David D. McPherson (Dept. of Internal Medicine, Univ. of Texas Health Sci. Ctr. at Houston, Houston, TX), and Christy K. Holland (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Xenon is a cellular protectant shown to stabilize or reduce neurologic injury in stroke. The goal of this work was to develop lipid-shelled microbubbles for ultrasound-triggered xenon release. Microbubbles loaded with either xenon alone (Xe-MB) or xenon and octafluoropropane (Xe-OFP-MB) were synthesized by high-shear mixing lipids with either 100% xenon, or 90% (v/v) xenon and 10% (v/v) octafluoropropane. The size distribution and the attenuation coefficient of Xe-MB and Xe-OFP-MB were measured using a Coulter counter and a broadband attenuation spectroscopy system, respectively. Gas chromatography/mass spectrometry (GC/MS) was performed to assess the dose and stability of encapsulated xenon. The feasibility of xenon release using 5-MHz pulsed Doppler ultrasound and 220-kHz pulsed ultrasound was tested by ultrasound attenuation spectroscopy. Co-encapsulation of OFP increased the number density, attenuation coefficient, and temporal stability of microbubbles. The GC/MS measurements revealed that 143 ± 20 μL/mg and 131 ± 33 μL/mg of xenon was loaded in Xe-MB and Xe-OFP-MB, respectively. Xe-MB and Xe-OFP-MB retained $54 \pm 11\%$ and $66 \pm 1\%$ of the xenon payload within 15 min of activation, respectively. Attenuation measurements confirmed ultrasound-triggered xenon release. These results suggest that lipid-shelled microbubbles with OFP could serve as ultrasound-triggered xenon delivery agents to attenuate cellular breakdown.

1:35

3pBAa3. Investigating the role of lipid transfer in sonoporation. Miles Aron, Oliver Vince, Michael Gray, and Eleanor P. Stride (Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, eleanor.stride@eng.ox.ac.uk)

The permeabilisation of cell membranes following exposure to microbubbles and ultrasound has considerable potential for therapeutic delivery. Recent studies have demonstrated that transfer of material takes place between phospholipid-coated microbubbles and cell membranes. The aim of this study was to investigate the impact of this transfer on membrane properties and cell permeability. Microbubbles were prepared with matched concentrations and size distributions from a series of formulations utilising phospholipids with differing molecular characteristics. Quantitative fluorescence microscopy techniques were used to quantify changes in the molecular packing, viscosity, and permeability to model drugs of cancer cells under mild ultrasound exposure (15 s of 1 MHz centre frequency, 125 kPa peak negative pressure, 1 kHz pulse repetition frequency, and 10% duty cycle). The molecular "packing" of cancer cell membranes was found to be

significantly affected by the transfer of both phospholipids and other microbubble coating constituents. This was particularly pronounced in the case of microbubbles containing a specific class of lipids, which promoted a ~4-fold increase in model drug uptake following ultrasound exposure. Similar results were observed in an investigation of permeability in an *in vitro* blood brain barrier model suggesting that microbubble composition may play a key role in the efficacy of ultrasound mediated delivery.

1:50

3pBAa4. Real-time imaging-guided microbubble mediated high intensity focused ultrasound heating in an *ex-vivo* machine-perfused pig liver. Dingjie Suo, Eric Juang, Sara Keller, and Michael Averkiou (Dept. of Bioengineering, Univ. of Washington, 616 NE Northlake Pl, Seattle, WA 98105, dsuo@uw.edu)

High intensity focused ultrasound (HIFU) is used clinically for the ablation of prostate and liver tumors. More recently, HIFU has been used in the brain for locally controlled thermal treatment of nonessential tremors. However, the thermal lesion location is restricted to a small area in the tissue due to geometric and boundary limitations (avoiding skin and bone burns). Bubble-enhanced heating (BEH) allows for the use of lower power and can generate spatially controlled thermal lesions in regions not currently accessible with conventional HIFU. We have studied BEH in vitro and in a machine perfused pig liver (MPL). The MPL presents a versatile platform for perfusion and BEH studies because it is clinically relevant and allows for delivery of microbubbles under ultrasound imaging. We first applied conventional HIFU and BEH in vitro, in an enclosure with acoustic windows contaiing glycerin and microbubbles while monitoring the temperature with a fine wire thermocouple. A focused 1 MHz transducer and various clinically approved microbubbles were used. Then we applied BEH on the MPL by introducing microbubbles under image guidance. With this initial study, we have observed that BEH has resulted in 5-10x energy reduction for the formation of a reference lesion in vitro, and 3-6x in the MPL.

2:05

3pBAa5. High-precision acoustic measurements of the non-linear dilatational elasticity of phospholipid-coated monodisperse microbubbles. Tim Segers (Phys. of Fluids Group, TechMed Ctr., Univ. of Twente, Enschede, Netherlands), Emmanuel Gaud (Bracco Suisse S.A., Geneva, Switzerland), Michel Versluis (Phys. of Fluids Group, TechMed Ctr., Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl), and Peter Frinking (Bracco Suisse S.A., Geneva, Switzerland)

The acoustic response of phospholipid-coated microbubble ultrasound contrast agents (UCA) is dramatically affected by their stabilizing shell. The interfacial shell elasticity increases the resonance frequency, the shell viscosity increases damping, and its nonlinear behavior increases the generation of harmonic echoes that are routinely used in contrast-enhanced ultrasound imaging. To date, the surface area-dependent interfacial properties of the phospholipid coating have never been measured due to the extremely short time scales of the MHz frequencies at which the microscopic bubbles are driven. Here, we present, high-precision acoustic measurements of the dilatational nonlinear viscoelastic shell properties of phospholipid-coated microbubbles. These highly accurate measurements are now accessible by tuning the surface dilatation of well-controlled monodisperse bubble suspensions through the ambient pressure. Upon compression, the shell elasticity of bubbles coated with DPPC and DPPE-PEG5000 was found to increase up to an elasticity of 0.6 N/m after which the monolayer collapses and the elasticity vanishes. During bubble expansion, the elasticity drops monotonically in two stages, first to an elasticity of 0.35 N/m, and then more rapidly to zero. Integration of the elasticity vs. surface area curves showed, indeed, that a phospholipid-coated microbubble is in a tensionless state upon compression, and that it reaches the interfacial tension of the surrounding medium upon expansion.

2:20

3pBAa6. Antibody targeted ultrasound-responsive nanodroplets for the therapy of brain metastases: A pre-clinical study. Oliver Vince, Luca Bau (Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. campus Res. Bldg., Oxford, Oxfordshire OX 3 7dq, United Kingdom, oliver.vince@worc.ox.ac.uk), Sarah Peeters (Oxford Inst. for Radiation Oncology, Univ. of Oxford, Oxford, United Kingdom), Michael Gray, Luke Richards (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), Sean Smart, Nicola Sibson (Oxford Inst. for Radiation Oncology, Univ. of Oxford, Oxford, United Kingdom), and Eleanor P. Stride (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom)

Metastatic tumors in the brain represent one of the leading causes of death from cancer with current treatments being largely ineffective and/or associated with significant side effects due to their lack of targeting. Conjugating MRI contrast agents with a monoclonal antibody for VCAM1 (anti-VCAM1) has previously allowed detection of brain tumor volumes two to three orders of magnitude smaller than those volumes currently detectable clinically. In this study, a novel magnetic and acoustically responsive phospholipid-stabilised nanodroplet formulation has been conjugated with anti-VCAM-1. Preliminary *in vivo* tests have shown that these anti-VCAM-1 nanodroplets can be successfully targeted to both inflamed areas of the brain and metastatic brain tumours. Acoustic droplet vaporisation of the anti-VCAM1 droplets, confirmed by passive cavitation detection, was also shown to cause blood brain barrier permeabilisation *in vivo*. Extensive *in vitro* characterisation of the potential of these nanodroplets to target brain metastases using antibody and magnetic targeting, along with the ultrasound conditions required for various therapeutic effects has been completed and has been found to agree well with mathematical modelling. The implications of these findings and the plans for future investigations into the therapeutic potential of these targeted and ultrasound responsive agents will be discussed.

2:35

3pBAa7. High-speed observations of Acoustic Cluster Therapy "activation" and "therapy". Jae Hee Song (CavLab, Medical and Industrial Ultrason., Univ. of Glasgow, Glasgow, United Kingdom), Andrew Healey (Phoenix Solutions, Oslo, Norway), and Paul Prentice (CavLab, Medical and Industrial Ultrason., Univ. of Glasgow, Ninewells Hospital, Dundee DD1 9SY, United Kingdom, paul.prentice@glasgow.ac.uk)

Acoustic Cluster Therapy (ACT^{TM}) is a novel and versatile drug delivery platform, based on electrostatically bound clusters of negatively charged contrast agent microbubbles (Sonazoid^{TM}, GE Healthcare) and positively charged oil microdroplets, for co-administration with drug to the vasculature. Preliminary *in-vitro* high-speed observations at up to 10 million frames per second of both "activation" (exposure of clusters to 2.19 MHz focused ultrasound, at mechanical index, MI $\approx$ 0.15) and "therapy" (exposure of the resulting vapour bubbles, to 675 kHz with MI ranging from 0.1 to 0.4) will be presented. At low MI, the therapy exposure induces quasi-linear oscillations of the vapour bubble, with higher MIs inducing high-order surface wave activity. In the latter case, the amplitude of the non-spherical oscillations increase during the exposure until involutions neck, generating intra-bubble droplets that traverse the bubble core at speeds of up to several m/s. Possible mechanisms for activation and therapy will be discussed, based on the observations. [This research was supported by ERC Starting Grant TheraCav, 336189, and Glasgow Knowledge Exchange project DefACT.]

Session 3pBAb

Biomedical Acoustics, Structural Acoustics, and Vibration and Physical Acoustics: Wave Propagation in Complex Media: From Theory to Applications IV

Guillaume Haiat, Cochair
Multiscale Modeling and Simulation Laboratory, CNRS, Laboratoire MSMS, Faculté des Sciences, UPEC, 61 avenue du gal de Gaulle, Creteil 94010, France

Pierre Belanger, Cochair
Mechanical Engineering, Ecole de technologie superieure, 1100, Notre Dame Ouest, Montreal, QC H3C 1K/, Canada

Contributed Papers

1:00

3pBAb1. Study on the ultrasonic wave convergence in the medium with bone. Masaya Saeki, Leslie V. Bustamante Diaz (Dept. of Elec. and Electronics Eng., Doshisha Univ., 1-3 Tatara-Miyakodani, Kyotanabe-shi, Kyoto 610-0321, Japan, ctwc0355@mail4.doshisha.ac.jp), Yoshiki Nagatani (Electronics Eng., Kobe City College of Technol., Kobe, Hyogo, Japan), Ko Chiba (School of Biomedical Sci., Nagasaki Univ., Nagasaki, Japan), Isao Mano (Dept. of Elec. and Electronics Eng., Doshisha Univ., Joyo, Japan), and Mami Matsukawa (Dept. of Elec. and Electronics Eng., Doshisha Univ., Kyotanabe, Japan)

Low intensity pulsed ultrasound can shorten the healing time of a bone fracture; however, the present technique does not seem not to consider the complicated ultrasound propagation in the body with bone. In this study, the time reversal technique was used to effectively converge ultrasound in the bone part. A three-dimensional distal radius model was constructed by using a high resolution-peripheral quantitative CT data (spatial resolution; 61 μm). In the simulation using Finite Difference Time Domain method, one cycle of sinusoidal ultrasound wave (1 MHz) with Hanning window was transmitted from a virtual fracture point (1 mm^3). The model was surrounded by water. The propagating ultrasound waves were observed by two ring shape array transducers, coaxially placed along the bone model at the distance of 30 mm from the fracture point. Then, time reversal waves were radiated from the array transducers, considering the phase shifts and amplitudes of the observed waves in the previous simulation. The time reversal waves successfully converged near the initial virtual fracture part, telling the usefulness of the technique. In the radial-tangential plane including the point, the converged area ($-$3dB of maximum stress value) was 42.8 mm^2.

1:15

3pBAb2. Evaluation of wave velocity in bovine meniscus by laser ultrasound technique. Takumi Fukunaga (Elec. and Electron. Eng., Doshisha Univ., 1-3, Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, ctwc0309@mail4.doshisha.ac.jp), Masaki Kuraoka, Taiki Makino (Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan), and Mami Matsukawa (Elec. and Electron. Eng., Doshisha Univ., Kyotanabe, Japan)

Laser ultrasound technique is suitable to measure the ultrasound velocity of small samples such as meniscus. In this study, longitudinal wave velocities in bovine meniscus samples were measured using a short pulsed laser. The samples were obtained from the knee joint of bovine femurs and five cubes ($3\times3\times3$ mm^3) were manually manufactured. The ultrasonic wave generated by the laser was propagated through the samples and the corresponding velocities in the axial, radial, and tangential directions were measured. The obtained velocity range was 2820–3560 m/s. The highest velocities were found in the tangential direction, followed by the velocities in the radial and axial directions, except for the samples extracted from the most posterior part of the meniscus, where the velocities were highest in the radial direction. In order to investigate the anisotropy of the meniscus, the cube samples were processed into octagonal prisms, and the velocities in the directions between tangential and axial directions were measured. The velocities observed in these directions were between the values of the ones in the tangential and the axial directions, showing the uniaxial anisotropy.

Invited Papers

1:30

3pBAb3. *In vivo* ultrasonic evaluation of distal part of radius in their early teens. Mami Matsukawa (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp), Isao Mano, Kaoru Horii (OYO Electric Co., Ltd., Joyo, Japan), Shiori Umemura, and Etsuko Ozaki (Kyoto Prefectural Univ. of Medicine, Kyoto, Japan)

For the evaluation of small radius of early teens, we have improved the ultrasonic bone measurement system, LD-100 (OYO electric)[1], which can evaluate the distal 5.5 % site of the radius of the non-dominant hand. The system measures two longitudinal waves (fast and slow waves) which propagate in the inside cancellous bone. We can also obtain the spatial distribution of wave attenuation during propagation in bone, which can be presented as an attenuation map. In our present teen's bone evaluation project (ages 12-18, n = 971), we found an open epiphysis at the distal end of radius in some attenuation maps of early teens. The area seems to be the growth plate, which have the ability to lengthen the bone by creating new cartilage within the bone itself. In case of boys, the plates were found

at the age of 14 or less. The values of body weights and heights of the teens with the growth plates were smaller than the average, telling that they are growing up. These results show the possibility of growth evaluation using ultrasound techniques. 1. H. Sai, *et al.*, Osteoporos Int. (2010) 21:1781–1790.

1:50

3pBAb4. Ultrasonic characterization of porous gyroid scaffolds for bone tissue engineering: Mechanical modelling and numerical validation. Giuseppe Rosi (MSME, Université Paris-Est, 61 Ave. du Géneral de Gaulle, Créteil 94010, France, giuseppe.rosi@univ-paris-est.fr), Vu-Hieu Nguyen (MSME, Université Paris-Est, Creteil, France), Adrien Loseille (GAMMA3 Team, INRIA, Saclay, France), and Salah Naili (MSME, Université Paris-Est, Créteil, France)

Bone substitutes can be used for pre-implant surgery in presence of volumetric bone defects. In this context, the ultrasound characterization of the bone substitute is a key issue. To this end, we model the implant as a 3D porous structure and we study its ultrasonic behavior. In the framework of artificial bone substitutes, microstructured scaffolds are widely used. In the literature, several geometrical configurations have been tested: among them, the gyroid-shaped scaffolds turn out to be an excellent choice, thanks to its ability to reproduce the behavior and the porous structure of trabecular bone. This study is focused on the mechanical modelling and numerical validation of wave propagation in a porous implant substitute. In particular, 3D finite-difference time-domain (FDTD) simulations were performed on a gyroid-shaped scaffold of saturated with water to validate the continuum mechanical model. Ultrasound excitations at different central frequencies were used in order to investigate the frequency-dependent behavior phase and group velocities.

WEDNESDAY AFTERNOON, 7 NOVEMBER 2018 ESQUIMALT (VCC), 1:00 P.M. TO 2:25 P.M.

Session 3pEA

Engineering Acoustics and Noise: Acoustic Particle Velocity Sensors, Algorithms, and Applications in Air

Michael V. Scanlon, Chair
RDRL-SES-P, Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783-1197

Invited Paper

1:00

3pEA1. Airborne vector sensor experiments within an anechoic chamber. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu), Peter H. Dahl (Acoust. and Mech. Eng., Appl. Phys. Lab., Univ. of Washington, Seattle, WA), W. C. Kirkpatrick Alberts, and Michael V. Scanlon (Acoust., Army Res. Lab, Adelphi, MD)

A technique to measure acoustic particle velocity in air is demonstrated in a controlled set of experiments conducted within an anechoic chamber at the Army Research Lab (ARL-Adelphi). This air-borne vector sensor measures acoustic pressure with an omnidirectional microphone and the 3D acceleration of a lightweight, rigid sphere with an embedded high-sensitivity tri-axial accelerometer. Two experiments are presented to demonstrate fundamental properties of the acoustic vector field: sound directionality, and the relative phase of pressure and particle velocity in the near to far field transition. Directional studies are implemented in the far-field of a speaker producing tones, defined by $kr>>1$, where r is the range from the speaker and k is the acoustic wavenumber. The pressure-velocity phase relationship is examined through the transition $kr << 1$, $kr = 1$, and then $kr >> 1$ by changing frequency and sensor range. For comparison, a tri-axial Microflown sensor simultaneously recorded the acoustic field. As neither sensor measures particle velocity directly, the phase characteristics of sensors depend on the sensing technique and corrections must be made to account for the true phase of particle velocity. Advantages and disadvantages of each method are discussed.

Contributed Papers

1:25

3pEA2. Localizing multiple targets with a single particle velocity sensor. Tung-Duong Tran-Luu and Minh Dao (US Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, tung-duong.tran-luu.civ@mail.mil)

The acoustic velocity sensor (AVS) measures the direction of a propagating plane wave by measuring instantaneously the particle velocity over a small volume in space. When multiple sound sources are present, the AVS no longer reports the correct directions as the sound intensity vectors (the product of pressure and velocity vector) get mixed up nonlinearly. A few methods have been proposed to extract the individual components from just one sensor by assuming, for example, spectral separability or nongaussianity. This paper shows it is in fact possible to extract the components without making such assumptions. The approach here is to constrain the signal with the acoustic impedance equation for multiple superposing plane waves and, applying various techniques, such as sparsity decomposition, to solve the derived least squares problem.

1:40

3pEA3. Seismo-acoustic detection and tracking of small aircrafts. Joydeep Bhattacharyya, Christian G. Reiff, Leng Sim, and W. C. Kirkpatrick Alberts (Sensors and Electronics Directorate, Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, joydeep.bhattacharyya.civ@mail.mil)

The Army Research Laboratory is undertaking an investigation to combine acoustic and seismic vector sensors, in conjunction with traditional acoustic arrays which use omnidirectional microphones, to detect and track small airplanes. The focus of this study is to improve the fidelity of the track with a sparse network and explore its applicability in both urban and rural settings. Towards this end, we carry out a field test where multiple aircrafts with known flightpaths are acoustically and seismically monitored over a period of several days. Using a detection algorithm that leverages both the blade frequency of the propellers and the Doppler associated with near-source propagation, we demonstrate that both acoustic and seismic data have the ability to detect and track the aircrafts. Both acoustic and seismic vector sensors have the ability to estimate both the arrival time and the azimuth to the source which lead to smaller sensor footprint. By combining multimodal sensing with novel target tracking techniques, we can reduce the impact of clutter while maintaining a robustness of our detection and tracking algorithms. A comparison of the acoustic and seismic vector sensors in tracking small aircrafts will be discussed.

1:55

3pEA4. Estimation of the source distance using the multiple three-dimensional acoustic intensimetry. In-Jee Jung and Jeong-Guon Ih (Mech. Eng., Korea Adv. Inst. of Sci. and Technol., 291, Daehak-ro, Yusung-gu, Daejeon 34141, South Korea, injee@kaist.ac.kr)

A 3D acoustic intensimetry (AI) in tetrahedral configuration can be used for the source localization. Such a 3D AI module is compact, but it can also yield a precise bearing angle for the low Helmholtz numbers if some bias errors are compensated. In this study, besides the bearing angle, the source distance is estimated by using the multiple 3D AI probes with different acoustic centers. Because each AI module indicates a vector from the acoustic center to the source direction, the nearest point between any pair of line vectors can approximate the source position. The source position can be estimated by minimizing the least square error of the positions determined by multiple intensity vectors. Such idea is preliminarily tested with four 3D AI modules, of which each module is composed of 4 MEMS microphones separated by 30 mm. Four 3D AI modules are also arranged in a tetrahedron layout, and the source frequency range is 0.5–2 kHz. The distance between modules is varied for 150–250 mm range, and the source distance for 1–4 m range. Current test results reveal that the estimation error is less than about 2, 5, and 7% when the elevation angle is within 0–30°, 30–60°, 60–90°, respectively.

2:10

3pEA5. Source localization using back propagation neural network and a single vector receiver. Wenbo Wang, Lin Su, Li Ma, and Qunyan Ren (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Haidian District, Beijing 100190, China, renqunyan@mail.ioa.ac.cn)

Matched field processing (MFP) has been widely used in source localization in shallow waters, whose performance is strongly correlated with the knowledge of environmental properties and proper selection of sound propagation model. This paper presents a neural network based approach for source ranging of moving target, which does not need heavy sound field calculation and no requirement of environmental information as a known prior. This neural network is designed to determine source range by observing multiple-frequency sound fields as excited by the source and recorded on a single vector receiver. In synthetic tests, this neural network is first trained on training data set as to adaptively select the optimal features through error back propagation, and then its localization performance is validated on testing data set. This approach is then tested on ship noise data as collected by a single vector sensor, and the relative error is smaller than 0.1 through comparing to GPS calculations. The promising results suggest the proposed approach can be further developed for source ranging applications with light system.

WEDNESDAY AFTERNOON, 7 NOVEMBER 2018

PREFUNCTION AREA 1B (VCC), 1:00 P.M. TO 2:00 P.M.

Session 3pEDa

Education in Acoustics: Undergraduate Research Exposition (Poster Session)

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

Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 2:00 p.m.

3pEDa1. Nonlinear tuning curve vibration using a column of non-wetted or wetted glass beads vibrating over a clamped elastic plate: Case for air-borne sound excitation. Emily V. Santos (Biological Sci., Univ. of Maryland Baltimore County, Phys. Dept., 572 C Holloway Rd., Annapolis, MD 21402, santosemily08@gmail.com) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

A soil plate oscillator (SPO) apparatus consists of two circular flanges sandwiching and clamping a thin circular elastic plate. The apparatus can model certain aspects of the nonlinear acoustic landmine detection problem which involves the interaction of granular material with an elastic plate in flexural vibration. Uniform spherical glass beads—representing a nonlinear mesoscopic elastic material—are supported at the bottom by the acrylic plate (4.5 inch diam, 1/8 inch thick) and stiff cylindrical sidewalls of the upper flange. Two 3 in. diam loud speakers placed above the bead column are driven with a swept sinusoidal signal applied to a constant current amplifier to generate airborne sound excitation from 50 to 1250 Hz. A small accelerometer fastened to the underside of the plate at the center measures the response. Separate tuning curve experiments are performed using a fixed column of 350 grams using 2,3,…,10 mm diameter soda-lime glass beads—wetted or non-wetted (mineral-oil). With dry beads, backbone curves (peak acceleration vs. corresponding resonant frequency) exhibit a linear region with comparable slopes, while detailed back-bone curvature vs. bead diameter reveals more structure. Wet beads exhibit less non-linear tuning curve behavior.

3pEDa2. Experiments in linear and nonlinear acoustic landmine detection. Francesca M. Browne and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m200726@usna.edu)

When nonmetallic land mines are difficult to detect using ground penetrating radar, acoustic landmine detection is used as a confirmation sensor. Non-metallic mines include VS 1.6 and VS 2.2 anti-tank plastic landmines. In modeling nonlinear acoustic landmine detection, a 20 cm diam drum-like landmine simulant (5.08 cm height, 0.64 cm thick aluminum bottom, 0.64 cm thick acrylic top plate) was used. The flexural vibration of the acrylic top plate with the soil behaves nonlinearly. A concrete soil tank filled with dry sifted masonry sand served as an idealized granular medium. The soil tank was 57.2 × 57.2 × 22.9 cm deep (wall thickness 15.2 cm). The simulant is 3 cm deep. In airborne excitation experiments, two 8 inch diam subwoofers radiate a swept tone. A geophone (1 inch diam) measured the rms particle velocity at 28 scan locations across a 56 cm segment on the surface. The geophone-microphone frequency response (between 50 Hz and 450 Hz) vs. scan position is used to extract a response profile at any individual frequency. The lowest drum-like mode shape was strongest at 156 Hz. Nonlinear tuning curve measurements at fixed locations exhibit more softening "on the mine" vs. "off the mine."

3pEDa3. Demonstration of acoustic landmine detection using a clamped soil plate oscillator with airborne sound excitation. Jenna M. Cartron (Phys. Dept., U.S. Naval Acad., 3519 Forest Haven Dr., Laurel, MD 20724, Jcartron11@gmail.com) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

Landmine detection in gravel road beds, soil, or sand for some mines (constructed out of plastic) are difficult to detect using ground penetrating radar. In acoustic landmine detection, one would like to remotely detect a vibration profile across the surface of the soil (sand or gravel) due to the interaction between a large soil column and the localized elastic top plate. In an idealized laboratory experiment, the soil plate oscillator (SPO) apparatus models an idealization of an acoustic landmine detection experiment. The clamped circular acrylic plate (1/8 in. thick) models the top plate of a plastic buried mine while the soil column (350 g dry sifted masonry sand) supported by the plate models the burial depth. A constant current amplified sinusoidal chirp from a sweep spectrum analyzer drives two three inch diam speakers (placed above the column) generating airborne excitation. A laser Doppler vibrometer (LDV) measured the rms particle velocity at 15 scan locations across the 4.5 inch diameter soil surface in sweeps between 50 Hz and 1250 Hz. The LDV-microphone frequency response vs. scan position is used to extract a response profile at any individual frequency. These mixed mode shapes are compared for both in-phase and out-of-phase excitation.

3pEDa4. Characteristics of the soundscape before and after the construction of the Block Island Wind Farm. Aditi Tripathy (Ocean Eng., Univ. of Rhode Island, Union Express, 50 Lower College Rd., Unit 4700, Kingston, RI 02881, atripathy@my.uri.edu), James H Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Jennifer Amaral, Kathleen J. Vigness-Raposa (Marine Acoust., Inc., Middletown, RI), Adam S. Frankel (Marine Acoust., Inc., Arlington, VA), Ying-Tsong Lin, and Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The Block Island Wind Farm (BIWF) south of Rhode Island is the first offshore windfarm in the United States. As part of the Ocean Special Area Management Plan, acoustic data were collected before the construction in the fall of 2009. Noise budgets were estimated based on this data and showed the dominant sources of sound in a 1/3-octave band centered at 500 Hz were shipping and wind. Data were again collected during and after construction of the wind farm and will be presented and compared to pre-construction levels. In 2009, Passive Aquatic Listener (PALs) were deployed. After construction was complete, data from a tetrahedral hydrophone array (~50 m from one of the wind turbines) were analyzed to study the soundscape from December 20, 2016 to January 14, 2017. The acoustic environment near the BIWF after construction showed contributions from shipping, wind, and marine life. Noise from the wind turbine was measured near 70 Hz at approximately 100 dB re 1 mPa at a range of 50 m. Significant marine mammal vocalizations were recorded including from humpback and fin whales. (Work supported by the Bureau of Ocean Energy Management.)

3pEDa5. Speech-language pathology student–clinicians' self-awareness of tongue position during rhotic sound production in American English. Megan Diekhoff and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

Proprioceptive signals from tongue positions are not strong, meaning it can be challenging to teach the articulation of speech sounds to language learners. Some of the most prevalent speech sound disorders in American English involve the rhotic /ɹ/. Speech-language pathologists provide modeling and feedback to their clients during treatment. Since tongue shape is not readily visible during rhotic production, clinicians rely heavily on verbal descriptions and auditory feedback to explain sounds to their clients. It is currently unclear how closely these approaches mirror or depend on veridical descriptions of tongue shape. An example from musical education provides an instructive backdrop: Although clarinet teachers frequently describe raising the tongue body for high notes, the tongue body must physically lower [Lulich *et al.*, 2017 JASA]. Students successfully learn to play high notes regardless of the accuracy of the description, but would clarinet instruction be more effective if it reflected physical reality? Analogous issues and questions can be raised in the context of speech-language pathology. This study presents preliminary results of an investigation of student-clinicians' self-awareness of tongue position during rhotic sounds.

Session 3pEDb

Education in Acoustics: General Topics in Acoustics Education

Benjamin V. Tucker, Cochair
Linguistics, University of Alberta, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada

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

Contributed Papers

2:30

3pEDb1. An Acoustical STEAM event for high school students. David A. Lechner (Mech. Eng., The Catholic Univ. of America, 9404 Bethany Pl., Washington, DC 20064, 66Lechner@cua.edu), Otto Wilson (Biomedical Eng., The Catholic Univ. of America, Washington, DC), and Shane Guan (Mech. Eng., The Catholic Univ. of America, Silver Spring, MD)

This presentation will review the methods and results of hosting our first Acoustical STEAM (Science, Technology, Engineering, Arts, and Math) competition as a part of the Washington DC Regional Chapter meeting of the ASA in the spring of 2018. Student teams were invited from over 50 Washington DC High School clubs and physics classes, provided a box assorted materials, and given 2 hours to create a musical video. The presentation will discuss the objectives, lessons learned, and results of the competition as well as student feedback on the event.

2:45

3pEDb2. A measurements laboratory built around a digitally-sampled microphone circuit. Andrew B. Wright (Systems Eng., Univ. of Arkansas at Little Rock, 2801 South University Ave., EIT 522, Little Rock, AR 72204, abwright@ualr.edu), Ann Wright (Phys., Hendrix College, Conway, AR), and Amanda Nolen (School of Education, Univ. of Arkansas at Little Rock, Little Rock, AR)

In Fall of 2017, a series of measurement laboratory exercises, built around a microphone preamplifier and filter, were used in teaching an engineering measurement techniques course. The microphone circuit included a preamplifier that introduced JFETs, a virtual ground and buffer to illustrate loading effects, a high pass filter and a low pass filter, and a zener-diode-based limiter. The data were sampled using a beaglebone microcontroller's built-in analog-to-digital converter. The data was transmitted to matlab for post-processing and analysis. The microphone was then used in the Spring 2018 acoustics class. A variety of projects, including a Kundt's tube, a driven closed-open pipe, a 3D printed horn for a blue-tooth speaker, and an absorptivity measurement for a restaurant's wall treatment tiles, were completed in the course. It was observed that students showed improved interest in the material (increased in-class and out-of-class questions about specifics of the course material related to the project, increased exploration of material outside the provided course materials, full participation by all members of the class in the project) and learning (improved test scores after the project began relative to scores before the project began) relative to earlier classes where no experimental component was included.

3:00

3pEDb3. Vibra-Son: A general audience exhibition dedicated to acoustics in Sherbrooke, Québec. Olivier Robin (Université de Sherbrooke, Université de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada, Olivier.Robin@USherbrooke.ca)

This presentation will present the ongoing setup of a general audience exhibition dedicated to acoustics in Sherbrooke, Québec. This project is built up with the Sherbrooke Nature and Science museum, Sherbrooke University (faculty and research groups) and with the contribution of external structures (like the Canadian acoustical association and companies). The exhibition will begin in 2019, June. The design of experiments that allows highlighting the interdisciplinary nature of acoustics is first described, as well as their setup in a limited space including possible travelling of the exhibition. The development or use of mobile applications to help sharing data or results with the visitors is especially under focus. The question of how including research projects into the exhibition is then discussed, with a dual objective of raising awareness of visitors about research but also making them a part of a research operation. Finally, satellites events that are currently planned are described, as well as their role in providing a multimodal description of acoustics, like a reading tour of comics in a library.

3:15

3pEDb4. Mathematica simulation of circular synthetic aperture acoustic imaging. Kathryn P. Kirkwood and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, m193342@usna.edu)

A Mathematica simulation will simulate circular synthetic aperture radar (CSAR) and its acoustic counterpart (CSAA). Two-dimensional and three-dimensional point targets will be imaged from echoes collected by a co-located transmitter/receiver moving along a circular track above the target plane. Previous work involved a Mathematica® (ver. 11) simulation of Synthetic Aperture Sonar (SAS). This simulation demonstrated how two-dimensional and three-dimensional point targets on a ground plane can be imaged from a collection of acoustic echoes using stripped-map geometry or a straight line track with a co-located transmitter/receiver element. The back-projection algorithm is used in both imaging geometries to construct a reflectance image of positioned point targets. The upcoming simulation will provide a comparison between circular track geometry and stripped-map straight line geometry.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

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

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Information retrieval from a soundscape by using blind source separation and clustering. Tzu-Hao Lin (Dept. of Marine Biodiversity Res., Japan Agency for Marine-Earth Sci. and Technol., 128 Academia Rd., Section 2, Nankang, Taipei 115, Taiwan, schonkopf@gmail.com), Yu Tsao (Res. Ctr. for Information Technol. Innovation, Academia Sinica, Taipei, Taiwan), Tomonari Akamatsu (National Res. Inst. of Fisheries Sci., Japan Fisheries Res. and Education Agency, Ibaraki, Japan), Mao-Ning Tuanmu (Biodiversity Res. Ctr., Academia Sinica, Taipei, Taiwan), and Katsunori Fujikura (Dept. of Marine Biodiversity Res., Japan Agency for Marine-Earth Sci. and Technol., Yokosuka, Japan)

Passive acoustic monitoring represents one of the remote sensing platforms of biodiversity. However, it remains challenging to retrieve meaningful biological information from a large amount of soundscape data when a comprehensive recognition database is not available. To overcome this issue, it is necessary to investigate the basic structure of a soundscape and subsequently retrieve biological information. The recent development of machine learning-based blind source separation techniques allow us to separate biological choruses and non-biological sounds appearing on a long-term spectrogram. After the blind source separation, the temporal-spatial changes of bioacoustic activities can be efficiently investigated by using a clustering algorithm. In this presentation, we will demonstrate the information retrieval in the forest and marine soundscapes. The separation result shows that in addition to biological information, we can also extract information relevant to weather patterns and human activities. Furthermore, the clustering result can be used to establish an audio library of nature soundscapes, which may facilitate the investigation of interactions among wildlife, climate change, and human development. In the future, the soundscape-based ecosystem monitoring will be feasible if we can integrate the soundscape information retrieval in a large-scale soundscape monitoring network.

1:25

3pID2. Yanny or Laurel? Acoustic and non-acoustic cues that influence speech perception. Brian B. Monson (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S Sixth St., Champaign, IL 61822, monson@illinois.edu)

"What do you hear?" This question that divided the masses highlights the complex nature of speech perception, which is dependent upon both acoustic and non-acoustic information available to the brain. In this talk, I will discuss some of the most recent advances in our understanding of both types of information that enable the complicated task of speech perception under adverse listening conditions. Recent research from our lab and others has revealed a surprising amount of speech signal information resides in the highest audible frequency range for humans (*i.e.*, beyond 8 kHz), providing localization, phonetic, and other speech source information. These cues, as well as non-acoustic (*e.g.*, experiential, visual, lexical) cues likely become increasingly important when a listener is faced with ambiguity and/or degradation of the acoustic signal. Given the diversity of prior experience from individual to individual, differences in perception of an ambiguous/degraded speech signal are to be expected.

1:45

3pID3. DJ Prof: Mixing instructional modes to improve student learning. Kathleen E. Wage (George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu)

Active learning encompasses a broad variety of activities designed to engage students with material while instructors are present to answer questions. In contrast, traditional lecture instruction is passive, and students' initial interaction with material is relegated to homework, which is often completed alone. Freeman *et al.*'s meta-analysis of 225 studies comparing lecture and active learning concluded that active learning is the "preferred empirically validated teaching practice" and showed that active learning courses have significantly lower failure rates than lecture courses [PNAS, 2014]. Incorporating proven active learning techniques into acoustics courses will improve student learning. In Acoustics Today [2016], John Buck, Jill Nelson, and I compared the professor in an active learning course to a DJ mixing different modes of instruction. DJ Prof designs a student-centered environment combining collaborative in-class exercises, short lecture segments, online video examples, and reading assignments. If you're an active learning skeptic, this talk will highlight empirical evidence of its benefits. If you're interested in getting started with active learning, this talk will suggest easy exercises to try. If you're an experienced DJ Prof, this talk will provide examples to amp up your pedagogical playlist.

Session 3pNS

Noise: Wind Turbine Noise

Kerrie G. Standlee, Chair
Daly-Standlee & Associates, Inc., 4900 SW Griffith Drive, Suite 205, Beaverton, OR 97005

Chair's Introduction—1:00

Contributed Papers

1:05

3pNS1. Simulated wind turbine emissions: Results from additional human response testing. Peggy B. Nelson, Matthew Waggenspack, Andrew Byrne (Dept of Speech-Language-Hearing Sci.,Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, peggynelson@umn.edu), Michael Sullivan (Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN), Meredith Adams (Otolaryngol., Univ. of Minnesota, Minneapolis, MN), and Jeffrey D. Marr (St. Anthony Falls Lab., Univ. of Minnesota, Minneapolis, MN)

Previous data (Nelson *et al.*, ASA 2017) indicated that healthy human adult participants experienced few symptoms from re-created wind turbine sound and infrasound emissions. That report included more than fifty subjects ages 21–73 years who attended to audible and infrasound signals generated from a wind turbine, recorded at 300 meters and re-created in a laboratory. Stimuli consisted of modulated and unmodulated audible turbine sound at 50 dB SPL, as well as natural and peak-enhanced turbine infrasound at an overall level of approximately 85 dB SPL (peaks up to 100 dB SPL). Participants were tested for their postural stability, detection, and ratings of audible and infrasound emissions randomly presented in one-minute exposure intervals in the laboratory. Very few and minor adverse effects had been noted to date, mostly ear fullness or pressure. Healthy participants showed no evidence of any change in postural sway in the presence of infrasound for the group tested. We have recently begun testing of participants who either a) live near turbines and complain of adverse effects, or b) who have symptoms of dizziness/imbalance as reported to their ENT specialist. Results from postural sway, sound quality judgments, and pre- and post-exposure symptoms will be reported from some of these participants. [Work supported by Xcel Energy RDF14.]

1:20

3pNS2. Audible noise sources (or characteristics?) from wind turbines using acoustic beam-forming. Andy Metelka (Sound and Vib. Solutions Canada Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

In many investigations of wind turbine noise, residents report specific and varying types of audible characteristics. With respect to the subject audible characteristics reported by residents (which are not picked in the dBA Leq measurements) turbines exhibit multiple audible sources at different operating and environmental conditions. Imaging techniques for real-time noise source identification have been used to identify tonal as well as dynamic pulsation during various environmental conditions and distances. Challenges with these techniques are presented using advanced imaging algorithms with successful imaging.

1:35

3pNS3. Existing aeroacoustic issues of building elements. Michael Bolduc (RWDI Air Inc., 600 Southgate Dr., Guelph, ON N1G 4P6, Canada, Michael.Bolduc@rwdi.com) and Andrew Bell (RWDI Air Inc., Copenhagen, Denmark)

During high wind events, certain building elements have the potential to generate significant tonal noise that can be heard miles away. These aeroacoustic issues are due to an interaction/coupling between an unstable fluid flow and a feedback mechanism, usually being a resonant acoustic mode. Aeroacoustic issues are often easy to avoid when susceptible geometry can be identified during the early design process, but mitigation can be technically difficult and very costly once issues do occur. Because aeroacoustic issues are unique and often annoying, they are often considered newsworthy and can be the subject of significant publicity. A series of case studies of existing structures, which have experienced aeroacoustic problems from their surrounding wind climate, are presented. Through each of these case studies, the steps used to identify suspect features and the recommended remedial measures are presented. The methodology for identification of the suspect feature may include investigating trends in weather data, conducting spectral analysis of audio recordings, performing calculations of the natural frequencies of building features, and on-site observations and measurements. Various mitigation strategies are detailed, depending on the mechanism found to be generating the aeroacoustic noise.

1:50

3pNS4. Wind turbine infra-sound penetration inside homes from multiple variable speed turbines. Andy Metelka (Sound and Vib. Solutions Canada Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Previous measurements in homes with respect to dominant turbines indicated the transmissibility factor calculations were a valid technique for describing wind turbine infra-sound penetration at various locations inside homes. However, when several turbines contribute at different RPM's and/or are not in phase different techniques need to be considered due to RPM variation and FFT densities. A comparison of the different methods is shown with examples of both cases.

Session 3pPP

Psychological and Physiological Acoustics: Pitch and Sound Localization

Ilse B. Labuschagne, Chair
Audiology and Speech Sciences, University of British Columbia, 4478 Haggart St., Vancouver, BC V6L 2H5, Canada

Contributed Papers

1:00

3pPP1. On the uncertainty principle for the pitch of brief tones. William M. Hartmann and Jon Pumplin (Phys. and Astronomy, Michigan State Univ., Physics-Astronomy, 567 Wilson Rd., East Lansing, MI 48824, hartmann@pa.msu.edu)

Many experiments have shown that the ability of human listeners to discriminate the frequencies of brief tones can violate the wave uncertainty principle that applies to linear systems. Several observations can be made: (1) Experiments that associate the duration of the tone envelope, or the second central moment of the squared envelope, with temporal uncertainty are consistent with the application of the energy-time principle in non-relativistic quantum mechanics. (2) The role of echoic memory in confounding the definition of duration can be avoided by immediately following the target tone to be judged by a tone of different frequency. (3) Whereas the uncertainty principle relates well-defined conjugate variables, such as time and frequency, listeners in psychoacoustical experiments may benefit from comparisons in both the pitch dimension (coding frequency) and a timbral dimension associated with the spectral distribution of energy. The latter provides additional information that is outside the context for the uncertainty principle. Its influence can be reduced by experiments in which the listener compares target tones having different brief durations. When these considerations are applied, it is found that human listeners still beat the uncertainty principle, but not as impressively.

1:15

3pPP2. Thresholds of noise in harmonic series maskers. Ilse B. Labuschagne and Valter Ciocca (Audiol. and Speech Sci., Univ. of Br. Columbia, 4478 Haggart St., Vancouver, BC V6L 2H5, Canada, ilse.labuschagne@alumni.ubc.ca)

Numerous studies have investigated the detection of pure tones and harmonic series in noise, but far fewer studied the perception of noise in harmonic series. One such study [Gockel, Moore and Patterson, J. Acoust. Soc. Am., **111**(6), 2759–2770 (2002)] demonstrated that the masking thresholds of noise in harmonic series maskers were affected by the fundamental frequency (F0) and the overall level of the series, and by the relative phase of the harmonic components. The maskers used by Gockel *et al.* comprised unresolved harmonics (10^{th} and higher harmonics below 5 kHz) for which F0 and frequency range covaried. The current study investigated noise detection for three non-overlapping spectral bands of equal auditory filter bandwidths (ERBs). Bands included either resolved harmonics (B1), unresolved harmonics (B3), or both resolved and unresolved harmonics (B2). Masker F0 and overall level were also varied. A Bayesian linear mixed-effects analysis showed that noise detection was better for higher F0s in B1 and B2, and that a higher presentation level resulted in a small improvement in noise detection in B2. Noise detection was better for higher presentation levels in B3. The findings will be discussed in relation to predictions of auditory processing models [Patterson, Allerhand, Giguère, J. Acoust. Soc. Am. **98**(4), 1890–1894 (1995)].

1:30

3pPP3. The sound image trajectory of a moving sound approaching a reflective wall. Shunya Kurachi, Daisuke Morikawa, and Tatsuya Hirahara (Dept. of Eng., Toyama Prefectural Univ., 5180 Kurokawa, Imizu, Toyama 939-0398, Japan, t754008@st.pu-toyama.ac.jp)

The movement of a sound image may not coincide with that of the sound source in a room with reflective walls. A listener sitting by a reflective wall perceives the movement of the sound image bending around their head when a sound source approaches the wall straight in the front of them. In order to investigate this perception quantitatively, we measured the trajectory of the sound image perceived by a listener against a sound source moving straight in front of them using a magnetic motion sensor. The sound source was a 1/3-octave band noise with a center frequency of 300, 750, 1500, 3000, or 6000 Hz emanating from a small moving loudspeaker at a sound pressure level of 70 dB. The loudspeaker moved on a rail perpendicular to the wall in front of the listener at a constant linear velocity. The listener tracked the direction of the moving sound image with his fingers by holding a motion sensor with his eyes closed. Results show that the sound image trajectory bent around the listener's head only when the stimulus contained low frequency components below 1.5 kHz, suggesting that interaural time difference plays an important role in the phenomenon.

1:45

3pPP4. An investigation of the training effects in a seven-week auditory localization training using augmented reality. Song Hui Chon and Sungyoung Kim (RIT, ENT-2151, 78 Lomb Memorial Dr., Rochester, NY 14623, sungyoungk@gmail.com)

We present the results from a seven-week auditory localization training using augmented reality with Microsoft HoloLens. For each trial, the target and distractor(s) would be randomly placed in space. In training, after the participant estimated the invisible target location, both true and estimated locations would be visually displayed. A localization score was calculated and recorded based on their distance. The test modules were identical to the training modules, except that the true location was not shown. The target was a female singing, and the distractor a piano accompaniment. Eight listeners were divided into two groups after the initial test, keeping the mean and variance of localization scores at the same level. Both groups showed a declining pattern in localization score over the eight tests, unlike the results from our previous study. The decreasing slope was smaller for the train group than the control group, which might reflect some mild effect of training. There was a one-time performance improvement after two trainings. The experiment might have been too simple to maintain participants' attention for weeks. Possible extraneous factors such as academic schedule might also have had an impact on this decreasing pattern.

2:00–2:15 Break

2:15

3pPP5. Adapting masking thresholds for spatially separated sounds in two dimensional ambisonics. Yuval Adler and Prateek Murgai (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., 660 Lomita Ct, Stanford, CA 94305, adler@ccrma.stanford.edu)

Spatial audio reproduction has gained renewed interest in recent years with the increasing presence of virtual reality applications. Ambisonics is currently the most widely adopted format for spatial audio content distributed via internet streaming services, and this has raised the need for audio compression relevant to the format. Current perceptual audio coders used for stereo audio content rely heavily on masking thresholds to reduce data rates, but these thresholds do not take into consideration spatial release from masking. This study begins an effort to update these thresholds for spatially separated sources in ambisonics. The listening tests were performed with sounds encoded in ambisonics to allow for tests to integrate whatever inherent limitations exist in the format. Initial listening tests were carried out for a subset of possible conditions—sound sources were separated along the horizontal plane, with a specific set of separation angles between the masker and maskee. Suggestions are given for continuing the work for the full range of possible conditions.

2:30

3pPP6. Right/left asymmetricity of horizontal sound localization of younger people. Tsukuru Osawa, Kazumoto Morita, Kenta Toyoda (Precision Mech. Eng., Chuo Univ., 1-13-27, Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan, tsukuru_osawa@camal.mech.chuo-u.ac.jp), Jo Sakashita, and Takeshi Toi (Precision Mech. Eng., Chuo Univ., Bunkyo-ku, 121-8551, Japan)

To investigate the performance of horizontal sound localization of younger people, authors conducted a test using 12 young participants (11 males, 1 females; ages 21–24.) The stimuli were presented through headphones to their right and left ears with 4 interaural time differences (ITDs) (0.2, 0.4, 0.6, and 0.8 ms) and 4 interaural level differences (ILDs) (3, 6, 9, and 12 dB(A)) besides the same condition to both ears. Concerning the frequency, pure tones of 0.5, 1.0, and 2.0 kHz, and composite sound of 0.5 + 2.0 kHz were presented, which lasted one second for each trial. The participants answered "Right" or "Left" after each trial according to their own judgment. The results of ITD condition showed that a right-sided inferiority was recognized in case of 1.0 and 2.0 kHz. Concerning the ILD condition, no right/left asymmetricity was shown. This asymmetricity may be caused by the improper setting of the experiment apparatus. Therefore, we switched right/left of the headphones to carry out an additional test in the same way. The same tendency as the original test was obtained, which means that the asymmetricity is likely due to the native characteristics of the participants.

2:45

3pPP7. Characteristics of horizontal sound localization of elderly people and analysis of its potential influential factors. Kazumoto Morita, Tsukuru Osawa, Kenta Toyoda, Jo Sakashita, and Takeshi Toi (Chuo Univ., 1-13-27, Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan, mtkkojiro@gmail.com)

To investigate the performance of horizontal sound localization of elderly people, authors conducted a test using 19 elderly participants (10 males, 9 females; ages 65-85; mean 71.5 years old.) The stimuli were presented through headphones to their right and left ears with 4 interaural time differences (ITDs) (0.2, 0.4, 0.6, and 0.8 ms) and 4 interaural level differences (ILDs) (3, 6, 9, and 12 dB(A)) besides the same condition to both ears. Concerning the frequency, pure tones of 1.0 and 2.0 kHz, and composite sound of 0.5 + 2.0 kHz were presented, which lasted one second for each trial. The participants answered "Right" or "Left" after each trial according to their own judgment. The results of ITD condition showed that in case of the pure tones a right-sided inferiority was recognized, which was poorer than the results of the younger participants. Meanwhile, no right/left difference was recognized with the composite sound. Concerning the ILD condition, no right/left asymmetricity was shown. Furthermore, no significant differences were recognized regarding the relationships between their sound localization performance and potential influential factors. The factors were Trail Making Test (TMT) results, their preferred ear, and their hearing loss difference between right/left ears.

WEDNESDAY AFTERNOON, 7 NOVEMBER 2018 SAANICH 1/2 (VCC), 1:00 P.M. TO 2:50 P.M.

Session 3pSA

Structural Acoustics and Vibration, Noise, and Signal Processing in Acoustics: History of Computational Methods in Structural Acoustics and Vibration

Micah R. Shepherd, Cochair
Applied Research Lab, Penn State University, PO Box 30, Mailstop 3220B, State College, PA 16801

Trevor W. Jerome, Cochair
Department of Acoustics, The Pennsylvania State University, GTWT Water Tunnel, State College, PA 16804

Invited Papers

1:00

3pSA1. Historical development of damping models in structural acoustics and vibrations. James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jgm@bu.edu)

This presentation discusses historical developments of damping models most commonly used in the analysis of structural acoustics and vibrations today. This presentation represents all damping models as frequency-dependent complex-valued matrices that appear in the equations of motion. The historical development of damping models has benefited from research on two fronts. The first is satisfaction of passivity and causality conditions by the damping model. The passivity condition requires no net power flow from the material over a cycle of vibration, while the causality condition requires that the material cannot respond before the excitation begins. The passivity condition is clearly stated in the frequency domain but is less obvious in the time domain. Conversely, the causality condition is clearly stated in the time domain but less obvious in the frequency domain. The second front is the affect of each damping model on computational methods used to evaluate the equations of motion. Discussion will include the use of undamped eigenvectors and Krylov spaces to create reduced-order models. Examples involving a rectangular plate will be presented to illustrate the features of commonly used damping model.

1:20

3pSA2. Determining acoustic sound power using vibration measurements and acoustic radiation modes. Caleb B. Goates (Dept. of Phys., Brigham Young Univ., Provo, UT), Cameron B. Jones (Mechanical Eng., Brigham Young Univ., Provo, UT), Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., N249 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu), and Jonathan Blotter (Mechanical Eng., Brigham Young Univ., Provo, UT)

In this paper, an efficient method for measuring the radiated sound power from consumer, industrial, and military products is presented. The method is based on spatially-dense vibration measurements from a 3D laser vibrometer and the acoustic radiation modes approach for computing sound power. The method will be a design tool that will allow engineers and designers to design quieter products by more effectively using noise as a design constraint. There are several standards that describe methods to compute sound power but these methods typically require specialized acoustic chambers for high accuracy results. The method presented here can be performed in an unknown acoustic environment without a reflective surface or an array of specific measurement points surrounding or half surrounding the noise source. The method will also be insensitive to varying background noise, temperature, and fluid/wind flow. The method will provide near real-time results and will be capable of measuring the radiated sound power from complex 3D geometries and built-up structures or parts of these structures. The method will presented in detail with experimental results validated the process.

1:40

3pSA3. Structural intensity on shell structures via a Finite-Element-Method approximation. Felipe Pires, Steve Vanlanduit, and Joris Dirckx (Biophys. & Biomedical Phys., Univ. of Antwerp, Groenenborgerlaan 171, Antwerp 2020, Belgium, felipe.pires@uantwerpen.be)

The study of energy transmission on plate-like structures is widely documented in literature. Its analysis requires the computation of the spatial derivatives of the out-of-plane displacement field of the studied sample and a priori knowledge of its material properties. However, if the structural intensity is to be assessed on irregular shells, such a study requires a more elaborate data processing. In addition to in-plane displacements, also the spatial derivatives along the sample's local coordinates are needed. For this purpose, a method was developed to approximate both the experimental displacement data and the spatial coordinates of a given arbitrary shell using a Finite-Element-Method model. After measuring the global displacement fields and their corresponding spatial coordinates for a given sample, the data was transferred to a shell model whose basis functions were properly defined in accordance to the application's demands and whose individual elements were assumed to behave in accordance with the Kirchhoff plate theory. The proposed method was able to process the experimental displacement and shape data of shells structures and proved to be a reliable tool to assess energy transmission.

2:00

3pSA4. A quasi-analytical formulation for acoustic radiation modes of simple curved structures. Caleb B. Goates, Scott D. Sommerfeldt, and Jonathan Blotter (Brigham Young Univ., N283 ESC, Provo, UT 84602, calebgoates@gmail.com)

Acoustic radiation modes have become a useful and widespread analysis tool in situations involving sound radiation from vibrating structures. They have found use in applications such as active structural acoustic control, optimization of structures for minimal sound radiation, and acoustical holography. Analytical expressions for the radiation resistance matrix, from which the radiation modes are obtained, are available only for a small number of simple source geometries, while the obtaining of radiation modes for more complicated structures typically requires BEM or similar computational methods. This paper details the development of quasi-analytical expressions for the radiation resistance matrices of singly-curved structures such as curved plates, cylinders, and angularly truncated cylinders. It is shown that the radiation modes of these structures may be obtained with a relatively low computational load compared to conventional methods.

Contributed Papers

2:20

3pSA5. On the physical meaning of subsonic wavenumber truncation as applied to source identification on vibration structures. Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu), Scott D. Sommerfeldt, and Michael T. Rose (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Wavenumber methods have been used to identify the supersonic portions of a vibrating structure that radiate to the farfield. To estimate the supersonic wave energy of a vibrating structure, the discrete Fourier transform can be used to determine the wavenumber spectrum which is then truncated above the acoustic wavenumber. The purely supersonic wavenumber spectrum can now be transformed back into the spatial domain to determine the vibration pattern associated only with the supersonic waves. Often, a cut-off coefficient associated with the acoustic wavenumber of the spatial radiation filter is used to reduce error. Equivalent spatial convolutions have also been formulated to obtain supersonic components of a vibrating pattern. This paper discusses the physical meaning of wavenumber truncation and its accuracy in identifying the surface areas of a vibrating structure that radiate sound. It is shown that truncating subsonic components of the vibration pattern will create non-physical vibration patterns which lie outside of the physical boundaries of the structure. Thus, subsonic wavenumber truncation methods may not be a reliable method for determining the radiated portions of a vibration pattern.

2:35

3pSA6. Experimental and numerical techniques for lightweight vehicle sound package development. Yuksel Gur (Res. and Adv. Eng., Ford Motor Co., Ford Res. and Innovation Ctr. (RIC), 2101 Village Rd., MD3135, Dearborn, MI 48121, ygur@ford.com)

The use of lightweight structural materials poses great challenge to noise control of a vehicle. A new approach to sound package design in lightweight vehicles was developed to reduce vehicle interior noise level without addition of weight. This new approach relies on lightweight acoustical materials that provide superior sound absorption performance to reduce noise level at the source, e.g., inside engine compartment and near the tires. Vehicle NVH finite element analysis (FEA) models were used to develop the lightweight vehicle's structural design to bring the lightweight vehicle's low and mid frequency responses closer the baseline vehicle's performance. Full vehicle SEA (statistical energy analysis) model was developed to evaluate the high frequency NVH (noise, vibration, and harshness) performance of a vehicle. This correlated SEA model was used for the vehicle sound package optimization studies to develop sound package design to improve lightweight vehicle's acoustic performance. In this paper, the use of NVH CAE (computer aided engineering) simulations for lightweight vehicle design is presented to highlight the limitation and use of FEA and SEA in lightweight vehicle development.

WEDNESDAY AFTERNOON, 7 NOVEMBER 2018 UPPER PAVILION (VCC), 1:00 P.M. TO 3:30 P.M.

Session 3pSC

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

Nadya Pincus, Chair
Linguistics and Cognitive Science, University of Delaware, 812 Lehigh Rd., Newark, DE 19711

All posters will be on display from 1:00 p.m. to 3:30 p.m. To allow authors an opportunity to view other posters in their session, authors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:15 p.m. and authors of even-numbered papers will be at their posters from 2:15 p.m. to 3:30 p.m.

Contributed Papers

3pSC1. English phoneme recognition of vocoded speech by adult Mandarin-speaking English-learners. Jing Yang (Dept. of Commun. Sci. and Disord., Univ. of Wisconsin-Milwaukee, Enderis Hall, 2400 E. Hartford Ave., Milwaukee, WI 53211, phial2000@gmail.com) and Li Xu (Commun. Sci. and Disord., Ohio Univ., Athens, OH)

The present study examined the joint influence of listeners' language experience and the degree of spectral degradation of speech signals on English phoneme recognition of L2 listeners. The participants included 27 native English-speaking listeners and 43 native Mandarin-speaking listeners who used English as an L2. The L2 participants varied in chronological age, age of English learning, length of residency in the United States, and the amount of daily-based English usage. The speech stimuli included 12 English vowels embedded in a /hVd/ context produced by four speakers and 20 English consonants embedded in a /Ca/ context produced by two speakers. The speech stimuli were processed using 2-, 4-, 6-, 8-, and 12-channel noise vocoders. The processed and original stimuli were presented to the listeners for identification in a random order. The results showed that spectral degradation had more adverse effects on the L2 listeners but the L2 disadvantage became more evident as the number of frequency increased. The L2 listeners showed different confusion patterns from the L1 listeners, which was affected by the L2 listeners' native language experience. Furthermore, the regression analysis revealed that the L2 listener's length of residency in the United States was a significant predictor for their phoneme recognition outcomes.

3pSC2. The lexical representation of second language length contrasts: Native English speakers learning Japanese. Rachel Hayes-Harb and Shannon L. Barrios (Univ. of Utah, 255 S. Central Campus Dr. Rm 2300, Salt Lake City, UT 84112, r.hayes-harb@utah.edu)

Adults experience difficulty using novel second language (L2) phonological contrasts to distinguish words. Indeed, even the ability to perceive and/or produce a novel contrast with relative accuracy does not guarantee an ability to implement the contrast to distinguish words in tasks that require lexical access. These observations lead to questions regarding the phonological content of learners' lexical representations of difficult L2 contrasts. We

employed an artificial lexicon study to examine the lexical encoding and implementation of Japanese consonant and vowel length contrasts by native English speakers. In the first experiment, native English speakers were taught a set of six Japanese-like auditory minimal pairs with pictured meanings. The members of each pair were differentiated by vowel length (e.g., [teki] vs. [teeki]). Participants were then asked to match the pictures to auditory words in four conditions: matched (see picture of 'teki', hear [teki]), vowel length mismatch (see 'teki', hear [teeki], and consonant length mismatch (see 'teki', hear [tekki]). Participants performed accurately on matched items, but were more likely to reject word forms mismatched for consonant than for vowel length. The results from this and subsequent experiments provide insight into the lexical encoding strategies used by learners for difficult novel contrasts.

3pSC3. Non-native perception of regionally accented speech of competing talkers. Ewa Jacewicz, Sasha Kim, and Robert A. Fox (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

Noisy conditions are challenging to non-native listeners who typically underperform relative to native listeners when attending to several competing talkers. We examine whether non-native listeners can utilize dialect-related cues in the target and in the masking speech, even if they do not reach the proficiency level of the native listeners. Our previous work with highly proficient Indonesian-English bilinguals (Fox *et al*., 2014) found that their performance differed markedly from native English listeners when the speech levels of the competing talkers were equal (0 dB SNR). We hypothesized that bilinguals cannot effectively separate utterances at 0 dB SNR due to a lack of a sufficient contrast in the voice levels of competing talkers, which may reduce their ability to benefit from the phonetic-acoustic details in order to "follow" a particular talker. The current study sought to replicate and validate earlier findings with a different group of bilinguals. The same experiment was conducted with Korean-English bilinguals. The results were replicated, providing further evidence that the ability to benefit from fine-grained phonetic details in regional accents declines for non-native listeners when the speech levels of the competing talkers are equal. Discussion will focus on defining the nature of the non-native speech processing deficit.

3pSC4. Effects of formant proximity and language experience on subcortical neural encoding of vowels in adulthood. Tian Zhao (Inst. for Learning and Brain Sci., Univ. of Washington, University of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu), Matthew Masapollo (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., Boston, MA), Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC, Canada), and Lucie Menard (Dept. of Linguist, Univ. of Quebec at Montreal, Montreal, QC, Canada)

Formant frequency convergence (or "focalization") and linguistic experience interact to shape the perception of vowels in adulthood. Here, we provide evidence of the effects of formant proximities and language experience at subcortical levels of the auditory pathway. Using a passive oddball/reversed oddball paradigm, the frequency-following response (FFR) in the auditory brainstem was elicited in sixteen healthy monolingual English speakers by a less-focal/English prototypic /u/ and a more-focal/French prototypic /u/. We examined the FFR as a function of stimulus type (English vs. French prototype) and condition (Standard vs. Deviant). A cross-correlation analysis revealed higher overall similarity between the FFR and evoking stimulus for the English prototype than the French prototype, suggesting an effect of language experience. Yet, deviants exhibited higher correlational values than standards. This effect was largely driven by the French prototype, suggesting an influence of focalization. A subsequent Fourier analysis revealed an effect of stimulus type at the *f0* peak, whereas only a condition effect was observed at the first harmonic peak. The latter effect was largely driven by the French prototype. Altogether, these findings suggest that subcortical encoding of vowels in adulthood is influenced by a complex interplay between formant proximities and long-term linguistic experience.

3pSC5. Effects of L2 experience on L2 vowel production and L1 speakers' perception of the improved vowels. Grace E. Oh (English Lit. and Lang., Konkuk Univ., 120 Neungdong-ro, Gwangjin-gu, Seoul 05029, South Korea, gracey1980@yahoo.com)

Effects of L2 experience on the production of L2 vowels were investigated and the perceptual accuracy of the L2 vowels by native speakers was tested to examine whether an improvement in vowel articulation leads to higher accuracy. A total of twenty Mandarin Chinese differing in the experience (6 months vs. 2 years) were compared to ten native Korean speakers in their production of seven Korean vowels, /i,ɛ,ɨ,ʌ,o,u,a/. More experienced Chinese speakers were expected to have acquired new Korean vowels, /ɛ,ɨ,ʌ,o/, in a more native-like manner than the inexperienced group. The results showed that Chinese learners were able to produce the three similar vowels, /i, u, a/ in a native-like manner even before any exposure to Korean. After 2 years of experience, they have shown to produce the new mid vowels, /ɛ, ʌ/ with greater height distinctions (F1 values) from adjacent high vowels. Although the two high back vowels, /o, u/ were deviant from the native norm, the Experienced group learned to distinguish the vowels with a distinctive F2 frequency, which has become a critical cue for the distinction of the two merging vowels by young Seoul speakers. When native Korean speakers were asked to identify the vowels produced by Chinese speakers, not only new mid vowels but also similar vowels were more accurately identified for the experienced group. The greatest improvement in perceptual accuracy was shown for the categories that were newly established by the inexperienced learners.

3pSC6. Foreign accent in L2 Japanese: A cross-sectional study. Kaori Idemaru (East Asian Lang. and Literatures, Univ. of Oregon, Eugene, OR 97403, idemaru@uoregon.edu), Kimiko Tsukada (Univ. of Oregon, Sydney, New South Wales, Australia), and Misaki Kato (Univ. of Oregon, Eugene, OR)

The current study examines acoustic sources of foreign accent in second language Japanese produced by American learners across different instructional levels and learning backgrounds. Our prior work has demonstrated that pitch accent, vowel duration, and spectral information of the vowel [e] influence perceived foreign accent in Japanese short sentences produced by intermediate learners, with pitch accent exerting the strongest influence. Building on this prior finding, the current study examines Japanese produced by American learners at the beginning level (n = 10), at the intermediate level (n = 16), American learners who have had early exposure to Japanese (n = 10), native Japanese speakers (n = 10), and ratings of the speech samples by native Japanese listeners (n = 10) to investigate the difference in the extent of perceived accent in their speech and the acoustic sources that influence perceived accent. The results of the current study shed light on issues related to development of second language speech, and the perceptual relevance of the development to lay listeners.

3pSC7. The role of voice familiarity in bilingual speech processing. Monika Molnar (Dept. of Speech-Lang. Pathol., Univ. of Toronto, 500 University Ave., Toronto, ON M5G 1V7, Canada, monika.molnar@utoronto.ca)

Interlocutor context affects proficient bilinguals' spoken language processing. For instance, bilinguals in a visual-auditory lexical decision task are able to predict the context-appropriate language based on the visual cues of interlocutor context (e.g., Molnar *et al*., 2015). Because it has been also demonstrated that bilinguals, as compared to monolinguals, process talker-voice information more efficiently (Levi, 2017), in the current study we addressed the question whether bilinguals are able to predict context-appropriate language based on voice information alone. First, in a same-different task, English monolingual and bilingual participants were familiarized with the voices of 4 female speakers who either spoke English (shared language across the monolingual and bilingual participants) or Farsi (unknown to both monolingual and bilingual participants). Then, in a lexical decision

task, the participants heard the same 4 voices again, but all of the voices spoke in English this time. We predicted that if the participants established a voice-language link in the first part of the task, then their response times should decrease when they hear an "English-voice" (as opposed to a "Farsi-voice") uttering an English word in the lexical decision task. Accordingly, our preliminary results suggest that the bilinguals' performance is facilitated by the established voice-language link.

3pSC8. Comprehension of accented sentences by young and older listeners. Yu-Jung Lin (Indiana Univ., 800 N Union St. Apt. 405, Bloomington, IN 47408-2230, lin41@indiana.edu)

Previous studies have shown that understanding accented speech requires additional cognitive support. Along this line, the current study aims to test whether the ability to understand accented speech can be influenced by aging and whether older listeners are more sensitive to accented speech. Native speakers of Southern Min from two age groups (18–35, 55 and older) are instructed to listen to sentences produced by two groups of speakers, native and nonnative speakers of Southern Min, answer questions regarding the target words in the sentences produced by these speakers, and then rate speakers' comprehensibility and accentedness. The stimuli for the listening test are produced by speakers during a delayed repetition task, where they repeat the Southern Min sentences after hearing each of them. The target words embedded in these sentences contain phonemes known to be difficult for nonnative speakers (e.g., syllable-initial voiced stops, nasalized vowels). It is hypothesized that older group, compared to younger listeners, will score lower on understanding accented speech, and tend to rate more speech accented and rate less speech comprehensible.

3pSC9. The effect of language experience on lexical tone perception. Xianzhuo Li (Int., College of Chinese Studies, Nanjing Normal Univ., Nanjing, Jiangsu, China) and Zhiyan Gao (English, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, zgao@gmu.edu)

Language experience has often shown to affect speech perception at segmental levels (Holt, 2010). Whether and how language experience affect perception of suprasegmental elements is often subject to dispute (Xu *et al.*, 2006; Peng, 2016). The current study focused on the identification and discrimination of Mandarin lexical tones by 3 groups of listeners, namely, 15 native speakers of Mandarin Chinese (tonal language), 15 native speakers of Lao (tonal language), and 15 native speakers of Uzbek (non-tonal language). We ask (1) whether experience with lexical tones (tonal vs. non-tonal) affects categorical perception of pitch contours; (2) whether differences in native tone inventories (Mandarin vs. Lao) affect pitch perception. Stimuli were 7 monosyllabic snippets resynthesized from a natural Mandarin speech sample. The stimuli represented a physical continuum of pitch contours ranging from a Mandarin level to a Mandarin rising tone. Preliminary results show a strong categorical perception of pitch contours for Mandarin speakers than for Lao and Uzbek speakers, and Lao speakers exhibited stronger categorical perception of pitch contours than Uzbek speakers. The results indicate that lexical tone perception is affected by both the experience with tonal languages and one's native tone inventory.

3pSC10. Perceptual cues used by Chinese native speakers in English /i/-/ɪ/ distinction. Zhuqing Wang (Ewha Womans Univ., Rm. 324, Inwunguawk, 52 Ewhayeodae-gil, Seodaemun-gu, Seoul, Seoul 03760, South Korea, jadeindurham@ewhain.net)

Native English speakers rely on spectral cues primarily and durational cues secondarily for /i/-/ɪ/ distinction whereas Chinese EFL learners tend to ignore tense-lax markedness due to a lack of lax /ɪ/ in Chinese vowel inventory and the speakers' insensitivity to formant values. Therefore, this study aims to explore whether Chinese speakers can use both durational and spectral cues for /i/-/ɪ/ distinction and whether females and males perform similarly in a perception test. The test was administered to 12 intermediate-level Chinese native speakers. *Beat* and *bit* were used as the basic stimuli and each stimulus was manipulated with seven different durations. The result showed that participants were predisposed to regard stimulus with long durations as /i/ and short durations as /ɪ/. Both the factor of gender and vowel duration had a significant effect on the correctness. Females were outperformed by males in the perception test correctness in most duration points except for very short vowels. Additionally, the size of differences between genders was larger in /ɪ/ than /i/ responses. In conclusion, although no significant spectral cue use was found with the participants, some uses of spectral cues at different vowel durations (90 ms, 390 ms, and 450 ms) were observed.

3pSC11. Exploring the skill of being good with accents: A study of monolingual English speakers' ability to reproduce a novel foreign accent. Laura Spinu (Dept. of Communications and Performing Arts, CUNY Kingsborough Community College, Brooklyn, NY) and Nadya Pincus (Dept. of Linguist and Cognit. Sci., Univ. of Delaware, 812 Lehigh Rd., Newark, DE 19711, npincus@udel.edu)

We collected production data from monolingual English speakers who were trained to produce a foreign accent of English, specifically Russian English (RE). First, the subjects read a paragraph in their own accent (baseline). Second, they listened to recordings of English sentences produced by a native speaker of Russian; next, the sentences were replayed, and the subjects imitated each sentence after hearing it. Finally, they were instructed to read the baseline sentences again (without audio prompts), trying their best to reproduce the RE accent. Several acoustic analyses are underway, addressing (a) vowel characteristics, (b) realization of stops, and (c) intonation patterns. Preliminary results show considerable differences in speakers' performance. The questions we address based on the data collected are (1) is there a continuous or more categorical change in speakers' ability to reproduce the RE accent? (2) which aspects of RE are most salient to native English speakers?, and (3) is there a correlation in quality and quantity such that speakers who pick up on more cues also produce them in more native-like manner? Our goal is to develop an algorithm to predict a given speaker's accentedness, and compare its performance with native speakers' accentedness ratings in a follow-up perceptual experiment.

3pSC12. Adult learners' use of lexical cues in the acquisition of L2 allophones. Shannon L. Barrios and Joselyn Rodriguez (Linguist, Univ. of Utah, 255 S.Central Campus Dr., Rm. 2313, Salt Lake City, UT 84108, s.barrios@utah.edu)

Adult second language (L2) learners gain knowledge of L2 allophones with experience. However, it is not well understood *how* this knowledge is acquired. We investigated whether naïve subjects use lexical cues in the form of visual referents to acquire L2 allophones. We exposed native English speakers to one of two artificial languages where novel words containing two acoustically similar sounds, [b] and [β], occurred in an overlapping distribution. The words were paired with an image that either did or did not reinforce the contrast (i.e., [bati]—"apple" and [βati]—"penguin" (*DiffImage* group) or [bati]—"penguin" and [βati]—"penguin" (*SameImage* group)). Participants completed three tests to determine whether the exposure phase impacted their ability to perceive and lexically encode the [b]-[β] contrast in trained and untrained words. If subjects use lexical cues in the form of visual referents to infer the phonological status of [b] and [β], then participants in the *DiffImage* group, but not the *SameImage* group, should discriminate and lexically encode the distinction. Data from 40 participants suggest that lexical cues may impact participants' lexical encoding, but not perceptual sensitivity, to the [b]-[β] contrast. We discuss our findings in relation to proposed learning mechanisms.

3pSC13. The determining factor of L2 vowel perception: The establishment of L2 categories. Wei Hu (State Key Lab. of Cognit. Neurosci. and Learning & IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., No. 241, Weijin Rd., Hexi District, Tianjin 300074, China, jykxhuwei@tjnu.edu.cn), Sha Tao (State Key Lab. of Cognit. Neurosci. and Learning & IDG/McGovern Inst. for Brain Res., Beijing Normal Univ., Beijing, China), Lihong Wang (School of Educational Sci., Tianjin Normal Univ., Tianjin, China), and Chang Liu (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

The L2 speech learning theories argue that the perception of L2 phonemes is mainly influenced by the relationship between and among L1 and L2 phonemes. In the present study, we hypothesized that L2 vowel identification is primarily determined by the solidity of the L2 vowel category establishment and the perceptual distance between L2 vowel categories. A group of 31 native Mandarin Chinese-speaking listeners and a control group of 9 native English-speaking listeners participated in the study. All listeners were asked to identify natural English/Chinese vowel identification and categorize and rate synthetic English/Chinese vowels. Major findings include: 1) Chinese learners' English vowel perception was associated with their capacity to establish English vowel category; 2) the perceptual distance between English vowel contrasts played a key role in Chinese learners' English vowel perception, rather than the distance between the Chinese vowel and English counterpart. Overall, results of this study suggest that the individual variability of Chinese-native listeners' English vowel perception is interpreted by the establishment of English vowel categories. [Work supported by China Natural Science Foundation 31628009.]

3pSC14. Production and perception of cross-linguistic categories by Polish-English heritage bilingual children. Maximilian Topps (Lancaster Univ., 6 Orchard Pl., Arundel BN18 9BP, United Kingdom, m.topps1@lancaster.ac.uk)

The acquisition of phonetic contrasts relies on acoustic cues such as duration and formant frequencies, which are used to distinguish acoustic signals. For second language (L2) learners, acquiring native-like L2 contrasts can be challenging when they are similar to first language contrasts, or are distinguished by different acoustic cues. Models such as the L2 Language Perception model (L2LP; Escudero, 2005), Perceptual Assimilation Model (PAM-L2; Best & Tyler, 2007), and the Speech Learning Model (SLM; Flege, 1995) make predictions regarding the outcomes of cross-linguistically similar contrast acquisition based on phonetic perception. This study investigates the similar contrasts of English /i/-/ɪ/ and Polish /i/-/ɨ/ acquired by English-Polish heritage bilingual children. The objective is to observe the models' predictions in the context of young bilinguals, and to compare strategies across perception and production. Data were collected from 18 children aged 5;11 to 10;7. For each contrast, a picture-naming task elicited production data, and a categorisation task, involving a phonetic continuum across a lexical minimal pair, elicited perception data. Results suggest that unique strategies are used in production for each language. However, this was not reflected in perception data, which suggests perceptual assimilation. Results are discussed within a framework of relative acoustic cue weighting.

3pSC15. A study on the vowel duration difference followed by phonetic contexts. Sooyoung Lee (English Lang. and Lit., Yonsei Univ., 207, 367-29, Seongsan-ro, Seoul, Seodaemun-gu 03726, South Korea, sree0910@naver.com) and Seok-Chae Lee (English Lang. and Lit., Yonsei Univ., Seoul, Sedeamum-Gu, South Korea)

Vowel duration in English serves as a cue to the voicing distinction of the following consonant, in a way that vowels preceding voiced consonants are longer than those preceding voiceless consonants. This study will examine the differences in vowel duration in various phonetic contexts produced by 8 Korean-speaking L2 learners of English in comparison with the differences produced by 4 native English speakers from the K-SEC corpus (Korean-Spoken English corpus). Specifically, we will test for the durational evidence of the influence of different manners of articulation and syllable positions on preceding or following vowels and investigate how the influence is phonetically realized by two language groups. Our hypotheses and predictions are as follows: 1) vowels are longer when preceded by consonants in coda position than when followed by the same consonants in onset position, 2) vowels are longer when preceded by nasals than when preceded by non-nasal consonants, and 3) Korean speakers produce the differences in vowel length to a smaller extent than native English speakers.

3pSC16. A multigenerational investigation of the acoustics of English-Modern Hebrew heritage speakers. Kyle Jones (Univ. of Arizona, 3015 Brunnet Ln., Sacramento, CA 95833, kysjones@email.arizona.edu)

This research investigates the speech acoustics of two generations of U.S. *olim* (immigrants) in Israel: First generation immigrants, whose first language (L1) is American English (AE), and their second-generation children, for whom English is a heritage language (HL). A specific HL accent has been demonstrated in the studies that have investigated the phenomenon, showing that heritage speakers have good control of phonetic/phonological contrasts between their two languages and distinct patterns from both NS and L2 learners (who both show L1 influence). The research focuses on issues of heritage language phonology and intergenerational multilingualism: What is the speech of HL speakers of AE in Israel like? How does this speech compare to the speech of their parents (their main source of input for AE)? How does Modern Hebrew (MH), their L2 or primary language, affect their AE? These questions are investigated through a language questionnaire and an acoustic analysis of voice onset time (VOT) in stops. Acoustic analysis demonstrates that HL speakers, echoing previous studies, have excellent control over phonetic and phonological contrasts in salient distinctions between their two languages, despite greater overall variability. VOT is closer to MH norms when speaking MH and closer to AE norms when speaking AE.

3pSC17. Perception of spectrally-shifted non-native speech. Michelle R. Kapolowicz (School of Behavioral and Brain Sci., The Univ. of Texas at Dallas, 800 West Campbell GR 4.1, Richardson, TX 75080, michelle.kapolowicz@utdallas.edu), Daniel Guest (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN), Vahid Montazeri (School of Behavioral and Brain Sci., The Univ. of Texas at Dallas, Richardson, TX), Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, Eugene, OR), and Peter F. Assmann (School of Behavioral and Brain Sci., The Univ. of Texas at Dallas, Richardson, TX)

Some studies have suggested that perceiving speech from multiple non-native talkers can be more difficult than perceiving speech from a single non-native talker. We found evidence in favor of this idea, previously showing a detrimental effect on sentence intelligibility of reducing the spectral resolution of non-native speech via a 9-channel tone vocoder (a condition where talker-specific spectral cues are limited). This effect was more severe when perceiving non-native speech from multiple talkers compared to a single talker, whereas we observed no differences across single- and multiple-native talker conditions. These results imply that listeners place greater reliance on talker-specific spectral cues to aid with perception of non-native speech as compared to native speech. In the present study, we further examined the effects of talker-specific spectral cues on intelligibility by scaling the spectral envelope and fundamental frequency (F0) of previously recorded sentences in successive increments of 8% and 30%, respectively, using the STRAIGHT vocoder. As predicted, intelligibility of spectrally-shifted sentences spoken from a single, non-native talker was similar to that observed for multiple non-native talkers, and listeners reported hearing spectrally-shifted speech as coming from different talkers. These results confirm the importance of spectral cues for the perception of non-native speech.

3pSC18. Lexical representation of Mandarin tones in second language learners. Kuo-Chan Sun (Univ. of AB, 3-20 Pembina Hall, University of AB, Edmonton, AB T6G 2H8, Canada, kuochan@ualberta.ca)

Previous studies have shown that a word's phonological similarity to other words (i.e., phonological neighborhood) can influence its recognition. However, most research concerning lexical representations has been observed for neighbors based on segmental overlap, and little is known about such effects with suprasegmentals such as Mandarin tones. In the present study, two experiments were conducted with forty L2 listeners and 40 native speakers to examine how tone neighborhood density influences Mandarin spoken word recognition. In Experiment 1, speed and accuracy

from both groups' performance in an auditory lexical task were influenced by tone neighborhood density (i.e., fewer words were recognized from dense tone neighborhoods than from sparse tone neighborhoods). However, L2 listeners' performance was inferior to native listeners'. In Experiment 2, form priming patterns showed that reliable facilitation was observed only when the prime and the target were identical, while monosyllabic Mandarin words differing only in tone failed to speed the response to the target. In addition, only L2 listeners showed an increase in RTs to respond to the target when it was preceded by tone overlap primes. The results of these experiments demonstrate that tone neighborhood is an important factor in L2 Mandarin spoken word recognition.

3pSC19. Effects of audiovisual perceptual training with corrective feedback on the perception and production of English sounds by Korean learners. Ho-Young Lee (Linguist, Seoul National Univ., Seoul, South Korea), Jookyeong Lee (English Lang. and Lit., Univ. of Seoul, Seoul, South Korea), Hyosung Hwang, and Joo Yeun Lim (Linguist, Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul 08826, South Korea, hshwang415@gmail.com)

This paper aims to propose a new audiovisual perceptual training with corrective feedback and compare the results of this training with those of audio only and audiovisual perceptual training. The results are compared in order to see which is more effective in improving the perception and production of English sounds by Korean learners. For this experiment, we selected 6 consonant pairs (i.e., /p-f/, /b-v/, /t-θ/, /l-r/, /s-ʃ/) and 4 vowel pairs (i.e., /iː-ɪ/, /ɛ-æ/, /ɑː-ʌ/, /ɔː-oʊ/), which are visually distinguishable from one another but often confused by Koreans. 20 subjects participated in audio only training (A group), 20 subjects in audiovisual training (AV group) and 25 subjects in audiovisual training with corrective feedback (CF group). 8 training sessions, each lasting about 40 minutes, were offered. All the groups showed a significant improvement in the perception of vowels and consonants (i.e., 5.4% and 3.4% improvements for the A group, 7.7% and 3.4% for the AV group, and 11.1% and 4.1% for the CF group). However, only A group showed a significant improvement in the production of vowels (i.e., 7.19% improvement) while only AV and CF groups showed a significant improvement in the production of consonants (i.e. 6.08% and 8.00% improvements for the AV and CF groups, respectively).

3pSC20. Listener flexibility to lexical alterations for foreign- and native-accented speech. Kali Burke (Psych., Univ. at Buffalo, SUNY, 246 Park Hall, Buffalo, NY 14260, kaliburk@buffalo.edu), Michelle K. Tulloch (Psych., Florida Atlantic Univ., Buffalo, NY), and Marieke van Heugten (Psych., Univ. at Buffalo, SUNY, Buffalo, NY)

Listeners regularly encounter speech produced by speakers born and raised in linguistic communities different from their own. Given that individuals from different linguistic communities often speak in a distinct fashion, it is not uncommon for listeners to be exposed to accent-induced variability. This can lead to discrepancies in the realization of words, rendering such speakers difficult to understand. As a result, listeners tend to increase their flexibility in their interpretation of variations in the phonological form for known words. To examine whether or not a foreign accent also leads to an increased flexibility in label-object mapping, we tested participants in a preferential looking paradigm. Using a between-subject design, participants either listened to a foreign- or to a native-accented speaker. In some trials, the target word labeled an object that was depicted on the screen (match trials), while in other trials the target word referred to an object that was closely related (but not an exact match) to one of the objects depicted on the screen (mismatch trials). Listeners' looking behavior after target word onset was examined across accents and trial types. The results of this study show that lexical alterations affect word recognition across foreign- and native-accented speech.

3pSC21. Production of English stop voicing distinction in familiar and unfamiliar positions by Spanish speakers. Alicia Swain and Chao-Yang Lee (Commun. Sci. and Disord., Ohio Univ., Grover Ctr. W252, Athens, OH 45701, as569911@ohio.edu)

The voicing distinction in English stop consonants involves distinct acoustic correlates depending on the position of the consonant. For Spanish speakers, prevocalic voicing distinction is in a familiar position whereas postvocalic voicing is in an unfamiliar position. In this study we examined whether Spanish speakers could use voice onset time and vowel duration to produce the voicing distinction in these two positions. Acoustic analysis showed Spanish speakers were able to produce the voicing distinction in both positions, although the magnitude of between-category difference is smaller compared to those produced by native English speakers. Production of the VOT distinction for prevocalic stops improved significantly following a brief tutorial on the acoustics of stop consonants, but the vowel duration difference for postvocalic stops did not show comparable change. These results suggest implementation of a phonological contrast in an unfamiliar position is constrained by native language characteristics. [Work supported by Ohio University College of Health Sciences and Professions Student Research Grant.]

3pSC22. Emotional responses to speech accommodation: A systematic review. Kathrin Rothermich, Havan Harris (Commun. Sci. and Disord., East Carolina Univ., 355 Beasley Dr., F3, Greenville, NC 27834, rothermichk17@ecu.edu), Kerry Sewell (Laupus Health Sci. Library, East Carolina Univ., Greenville, NC), and Susan Bobb (Psych., Gordon College, Wenham, MA)

Five percent of the U.S. population speak English "not well" or not at all, often leading to miscommunication, which can be especially problematic in a healthcare environment or at the workplace. According to socio-linguistic frameworks such as Communication Accommodation Theory, native speakers have interactional goals and strategies to communicate effectively with non-native speakers. One of these strategies, foreign-directed speech, involves modifying the speech output acoustically and linguistically by slowing down, simplifying speech, and exaggerating vowels. While the acoustic properties of foreign-directed speech are well documented, research is limited on how non-native speakers interpret them emotionally and pragmatically. For instance, do non-native speakers find foreign-direct speech helpful or does it come across as condescending? The purpose of this systematic review is to determine the current evidence on how speech accommodation affects the listener, with a special focus on how it is perceived by non-native speakers. In this review, we outline the basic components of communication accommodation, provide a systematic review of the literature related to the emotional effects of speech accommodation, and discuss current issues. We conclude by formulating recommendations for future research.

3pSC23. Acquisition of categorical perception of Mandarin tone. Michelle X. Li (Office of Linguist, Univ. of Victoria, PO Box 1700 STN CSC, Victoria, BC V8W 2Y2, Canada, xiao.xiao.li@mail.mcgill.ca), Daniel Rivas (Psych., Université du Québec à Montréal, Montreal, QC, Canada), Fernanda Pérez Gay Juárez (Neurosci., McGill Univ., Montreal, QC, Canada), Tomy Sicotte, and Stevan Harnad (Psych., Université du Québec à Montréal, Montreal, QC, Canada)

This experiment examined the effect of motor repetition on the learning of Mandarin pitch categories by non-native speakers. In Mandarin, there are four tonal categories, which refer to pitch contours that discriminate words. Differentiating tonal categories is both essential and arduous for non-native speakers to learn. Native speakers do this effortlessly because they have a categorical perception (CP) effect for tones, i.e. they perceive items within a tonal category as more similar to each other and items between categories

as more different. Non-native speakers do not have this effect, and this experiment attempted to induce the CP effect for Mandarin tones via a training task, which was the experimental manipulation: Participants either repeated a tone stimulus before categorizing it or listened to a stimulus and categorized it without repetition. Discrimination between and within tonal categories was measured before and after training. All participants demonstrated increased between-category and within-category discrimination after training, except for learners who repeated stimuli in the training phase. They demonstrated a decrease in within-category discrimination, showing a weak CP effect that could be stronger with more training. Implications of these results on auditory category learning and language education will be discussed.

3pSC24. Effects of the place of articulation of the following consonant on the identification and discrimination of American English vowels by native speakers of Japanese and Korean. Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp) and Heesun Han (Osaka Univ., Toyonaka, Osaka, Japan)

Native speakers of Japanese and Korean identified 6 American English monophthongs /i, ɪ, ɛ, æ, ɑ, ʌ/ and discriminated 6 vowel pairs /i/-/ɪ/, /ɛ/-/ɪ/, /æ/-/ɛ/, /æ/-/ɑ/, /æ/-/ʌ/, /ɑ/-/ʌ/ in in /hVt/, /pVt/, /pVn/ and /pVl/ frames. The two groups of listeners' identification and discrimination accuracy were submitted to 2 mixed-design ANOVAs, respectively, with 2 Listener Groups as a between-subject variable, and 6 frames and 6 vowels (or vowel pairs) as within-subject variables. As for identification, a main effect of vowel and frame are both significant ($p<.001$), and a three-way interaction of listener groups × vowels × frames is also significant ($p<.001$). Post-hoc pair-wise comparisons revealed that Korean listeners identified /ʌ/ significantly better than Japanese listeners. This is probably because Korean listeners can equate the English vowel to Korean /ʌ/. Japanese listeners' identification accuracy of /i/ is lower before /n/ and /l/, but that of Korean listeners was not affected by the context. As for discrimination, a main effect of the frames and vowel pairs was significant ($p<.001$). A three-way interaction of the 3 factors was not significant, but all the two-way interactions were significant ($p<.001$). Post-hoc pair-wise comparisons revealed that Japanese listeners' discrimination accuracy of /i/-/ɪ/ and /æ/-/ɛ/ is lower before /n/ and /l/ than before /t/, while that of Korean listeners was not affected by the context. Korean listeners discriminated /æ/-/ɛ/ least accurately of all the six vowel pairs.

3pSC25. Information-theoretic variables in Spanish-English bilingual speech. Khia A. Johnson (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada, khia.johnson@alumni.ubc.ca)

Languages lenite similar segments to different extents—a finding that can be accounted for with information-theoretic variables like frequency, predictability, and informativity [Cohen Priva, 2017, Language 93: 569–597]. Prior research addresses the role of segment information content across languages, but assumes that information-theoretic variables operate on an in-language basis. While appropriate for monolingual speech, this assumption is problematic for bilingual speech, as an individual's languages are known to influence one another (e.g. [Fricke *et al*., 2016, J. Mem. Lang. 89: 110–137]). In this paper, I report on a study using the Bangor Miami corpus [Deuchar *et al*., 2014, Advances in the Study of Bilingualism: 93–110], addressing how well information-theoretic variables predict lenition in Spanish-English bilingual speech. Specifically, do they operate on an in-language or cross-language basis? To address this, I focus on the duration of word-medial intervocalic fricatives shared by both languages—/f/ and /s/. Accounting for variables known to affect segment duration (e.g. speech rate), I use linear mixed-effect models to assess the contribution of information-theoretic variables in accounting for duration variation in bilingual speech. Four models are evaluated, compared, and discussed: (i) English in-language, (ii) English cross-language, (iii) Spanish in-language, and (iv) Spanish cross-language.

3pSC26. The effect of native language on the second language vowel variability. Jaekoo Kang (Speech-Language-Hearing Sci. program, CUNY Graduate Ctr., 3547 34th St., Apt. 1E, Long Island City, NY 11106, jkang@gradcenter.cuny.edu), D. H. Whalen (Speech-Language-Hearing Sci. program, CUNY Graduate Ctr., New York, NY), and Hosung Nam (English Lang. and Lit., Korea Univ., Seoul, South Korea)

Korean learners of English must create four vowel categories for English (/i, ɪ/ and /ɛ, æ/) in relation to two similar native categories (/i/ and /ɛ/). It is hypothesized that new categories should be easier to learn than similar ones (Flege, 1994), but it is unclear whether the English L2 vowels are similar or new. The degree of similarity between the four English vowels and the two Korean vowels was examined using the distribution metrics (i.e., ellipse overlap, cross-entropy, and Gaussian Mixture Model) as well as Euclidean distance in F1/F2 space. The L2 spoken corpus included 100 repetitions of words in both Korean and English spoken by 37 Korean L2 learners (20 female). Preliminary results indicate that the English (L2) high vowel pair was more overlapped with Korean /i/ than the low vowel pair with Korean /ɛ/, especially for male speakers. For the English (L2) low vowel pair, female speakers showed less overlap but higher variability along F1 direction than male speakers. This demonstrates that the similarity between Korean and English vowels is characterized by the distribution as well as the distances between the vowel categories. Acoustic results will be further compared with identification by native English speakers.

3pSC27. Difference of articulatory movement between native and non-native consonant clusters. Seiya Funatsu (Sci. Information Ctr., Prefectural Univ. of Hiroshima, 1-1-71 Ujinahigashi Minami-ku, Hiroshima 734-8558, Japan, funatsu@pu-hiroshima.ac.jp) and Masako Fujimoto (Waseda Univ., Tokorozawa, Saitama, Japan)

We investigated the difference of articulatory movement between native and non-native consonant clusters. English has consonant clusters, but Japanese does not. Therefore, speakers chosen for comparison were native English and native Japanese speakers. Speech samples consisted of 4 words, "blat," "bnat," "plat," "pnat." In these words, /bl/ and /pl/ are English clusters, but /bn/ and /pn/ are not. We measured the movement of the tongue tip, the mandible and the lower lip by WAVE system (NDI corp.). There were remarkable differences in the mandible and the lower lip movement between native (/bl/, /pl/) and non-native (/bn/, /pn/) clusters in English speakers. Namely, with the non-native clusters the difference of the articulatory movement in the mandible and the lower lip of every utterance was quite large; however, in native clusters, the difference was quite small. For Japanese speakers it was large for all clusters. Thus, it was revealed that the articulatory movement of the mandible and the lower lip in non-native clusters was not stable in native English speakers, even though English has consonant clusters. (This study was supported by KAKENHI 15K02524, 17K02707.)

3pSC28. An ultrasound investigation of how accurately people follow tongue movement instructions. Ian Wilson (CLR Phonet. Lab, Univ. of Aizu, Tsuruga, Ikki-machi, Aizuwakamatsu, Fukushima 965-8580, Japan, wilson@u-aizu.ac.jp) and Julián Villegas (Comput. Arts Lab., Univ. of Aizu, Aizu Wakamtsu, Fukushima, Japan)

In an analytic-linguistic approach to teaching segmental pronunciation and articulatory setting, teachers and voice coaches give explicit instructions on tongue placement and movements. Instructors assume that learners can do exactly as instructed. This assumption was tested in research by Wilson and Horiguchi (2012, PSLLT), who showed that phonetically untrained participants were very poor at following explicit tongue movement instructions. In their study, both the magnitude and direction of movement of the tongue's centre of gravity were calculated from 2D ultrasound images. However, by only measuring changes in the centre of gravity, it is possible that movements were found to be smaller than they really were, especially if participants focused on the front of the tongue rather than the whole tongue body. In this study, we reanalyzed the original data, this time focusing on the surface of the tongue, rather than the centre of gravity. We made tongue traces using EdgeTrak software (Li *et al.*, 2005), and compared them using a Smoothing-Spline ANOVA method (Davidson, 2006). Results differed from the original study, showing that participants were more conscious of what the front of the tongue was doing rather than the whole tongue body. Implications for segmental pronunciation teaching will be discussed.

3pSC29. The effectiveness of audio-visual training on non-native English speech production and perception. Chia-Ni Shen (Linguist, Univ. of Oregon, 17460 SW 104th Ave., Tualatin, OR 97062, jshen@uoregon.edu) and Melissa M. Baese-Berk (Linguist, Univ. of Oregon, Eugene, OR)

This project examines the effectiveness of audio-visual training on non-native English speech production and perception. Previous research utilizing audio-visual training has been employed in the field of speech pathology, showing positive outcomes in improving speech among dyslexic children. However, few studies to date have examined its use in second language learning, specifically bilabial and labiodental consonants (i.e., /b, p, m, f, v/), which are known to be challenging for many second language learners. The aim of this project is to explore audio-visual training across 3 native language groups, Mandarin Chinese, Japanese, and Arabic, who are all English language learners. Participants undergo a training regimen designed to examine the effects of audio-visual and audio-only training. Performance before and after training is assessed via perception and production tests. We hypothesize that 1) production and perception performance will improve after two days of training and will improve more after audio-visual training and 2) production and perception improvement will rely heavily on a participants' language background and known difficulties with specific bilabial and labiodental sounds. Results from this study will enrich understanding of the effectiveness of audio-visual training in second language learning.

3pSC30. Parallel adjustment of phonetic targets in L2 English voice onset time. Eleanor Chodroff (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, eleanor.chodroff@northwestern.edu) and Melissa M. Baese-Berk (Linguist, Univ. of Oregon, Eugene, OR)

The phonetic realization of any given speech sound varies considerably across speakers and languages. For instance, the voice onset time (VOT) of [k^h] can range from ~40 ms to over 100 ms (Cho & Ladefoged, 1999). Within American English, recent research has demonstrated that the VOT of [k^h] is also systematically and linearly related to the VOTs of [p^h] and [t^h] (Chodroff & Wilson, 2017). In the present study, we investigated whether these relations were maintained across L2 speakers of English. L2 speech production could arise from independent acquisition (or adjustment) of the phonetic targets for each speech segment. Alternatively, the presence of covariation would indicate that properties of speech sounds may be altered in tandem. To investigate this, VOT data was obtained from a subset of the ALLSSTAR Corpus, which contained matched connected speech data from 26 L1 American English speakers and 114 L2 English speakers (22 unique L1s). Preliminary analysis revealed strong VOT covariation among aspirated stops across L2 speakers (rs = 0.67 to 0.74), and qualitatively equivalent covariation across L1 speakers (rs = 0.57 to 0.75). The observed covariation may arise from a universal principle of uniformity requiring near-identical implementation of the shared laryngeal feature value.

3pSC31. A study of the difference vowel duration according to pronunciation assessment and the stress. Kihoon Park (Yonsei Univ., Yonsei University Graduated School of Education, Seoul, Seodaemun-gu 120-749, South Korea, manim0317@naver.com) and Seok-Chae Lee (Yonsei Univ., Seoul, Sedeamum-Gu, South Korea)

The purpose of this paper is to examine Korean elementary English learners' production of lexical and phrasal stress and the difference of the vowel length according to stress and pronunciation assessment. For the study, Korean-spoken English Corpus spoken by Korean elementary students was used. Lexical and phrasal stress were analyzed in terms of vowel length. As a result of the analysis, the following was revealed. A) Higher level groups of pronunciation assessment pronounced stressed syllables much longer than unstressed syllables compared to the lower level groups. B) As a result of the comparison according to the stress position, it was found that the stressed vowel located in the second syllable was pronounced much longer in all the pronunciation assessment group. It can be predicted as a result of the effect of mother tongue interference, which does not allow diphthong. C) Speakers that produced stressed syllables with much longer duration than unstressed syllables were classified as higher level groups in assessment. The result of this research suggests that segments and supra segments should be taught at the same time.

3pSC32. Quantifying fine-scale details of vowel spaces in German language learners. Lauren Elliott (Psych., Carthage College, 2001 Alford Park Dr., Kenosha, WI 53140, lelliott1@carthage.edu) and Benjamin N. Taft (Landmark Acoust. LLC, Racine, WI)

Second language learners may be particularly challenged when the new language makes distinctions between sounds that are phonetically equivalent in the speaker's first language. If the distinction occurs between vowels, native speakers and learners should occupy the relevant vowel spaces in different ways. Native speakers' vowel spaces should show denser, more acoustically separate regions for each vowel. Learners' spaces should show less consistent, more overlapping spaces. In contrast, when a pair of vowels are phonetically distinct in both languages, there should be little difference between native and learner acoustic spaces. We use a custom formant tracker to quantify the vowel spaces of two native speakers and 6 German language learners.

3pSC33. Digit span error patterns in bilinguals and monolinguals. Noah M. Philipp-Muller (Linguist, Univ. of Toronto, 4th Fl., 100 St. George St., Toronto, ON M5S3G3, Canada, noah.philippmuller@gmail.com), Laura Spinu (Communications and Performing Arts, CUNY, New York, NY), and Yasaman Rafat (Modern Lang. and Literatures, Western Univ., London, ON, Canada)

Research shows that bilinguals tend to outperform monolinguals on certain cognitive and linguistic tasks. While the mechanism underlying these advantages remains unclear, it has been suggested that bilinguals have enhanced working memory, which may be responsible for some of the cognitive advantages observed in these populations. To examine the mechanism responsible for the bilingual advantage, serial working memory was compared between monolingual and bilingual undergraduate students using an adaptive digit span test ($n = 77$). The results of the test were algorithmically adjusted in order to reveal not only if each digit was correct, but also the existence of serial errors and digit scrambling. The results showed that bilinguals made significantly fewer transpositional errors compared to monolinguals ($p < 0.02$, $d = 0.61$). These results are thought to be caused by differences in attention to serial information between bilinguals and monolinguals. Another possible explanation for these results is that bilinguals implement more cognitive memory techniques (like chunking), and are therefore better at retaining serial information. The computational methods developed in this experiment will help guide paths for further research on the impact of syntactical co-activation on serial memory recall, and primacy/recency effects.

Session 3pSPa

Signal Processing in Acoustics: General Topics in Signal Processing I (Poster Session)

Kay L. Gemba, Cochair
MPL/SIO, UCSD, University of California, San Diego, 8820 Shellback Way, Spiess Hall, Room 446, La Jolla, CA 92037

Jit Sarkar, Cochair
Marine Physical Laboratory, Scripps Institution of Oceanography, 9500 Gilman Drive, Mail Code 0238, La Jolla, CA 92093-0238

All posters will be on display and all author will be at their posters from 1:00 p.m. to 3:30 p.m.

Contributed Papers

3pSPa1. Assessing multimodal speech-gesture coordination: correlation map analysis for signal comparison. Samantha G. Danner (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, sfgordon@usc.edu)

Speech acoustics and body movements are systematically linked during speaking; however, there is no consensus regarding which acoustic and/or kinematic signals are relevant for this coordination nor what critical landmarks facilitate this linkage. To address this, we use correlation map analysis (CMA) of speech acoustics and body movement to assess the hypothesis that speech and gesture are coordinated at prosodically prominent speech regions. CMA allows for the analysis of correlation between any two continuous signals both instantaneously and at user-defined ranges of delay. In this study, we examine the relationship of the speech's RMS acoustic amplitude signal with the dominant hand's velocity signal at speech turns, phrase boundaries, acoustic amplitude peaks, and other ("elsewhere") regions. We find that turns and phrase edges exhibit the greatest likelihood of positive correlation. Additionally, the likelihood of correlation is higher when the hand velocity signal is delayed with respect to the amplitude signal. These results suggest that speech and gesture may be strongly linked at speech turn landmarks for the purpose of signaling a floor exchange. Thus in addition to coordination for semantic and prosodic purposes, we propose that speech-gesture coordination can serve with prosody to signal speech turns in conversation. [Work supported by NIH.]

3pSPa2. Compressed sensing and recovery of underwater acoustic signal in Internet of Things. Feiyun Wu, Kunde Yang, Quan Sun, and Yunchao Zhu (Northwestern PolyTech. Univ., Youyi Rd 127, Xi'an, Shaanxi 710072, China, wfy@nwpu.edu.cn)

Telemonitoring and communications technologies of underwater information is inseparable from the process of sampling and transmitting of data. Conventional methods often fail in energy efficiency to obtain such a huge data in Internet of Things (IoT), compressed sensing (CS) provides a new perspective to solve the problem. Unfortunately, the underwater acoustic signal is non-sparse in the time domain and thus the current CS methods cannot be used directly. This study adopts the fast Fourier transform (FFT) based dictionary-matrix for sparse representation. Then we design an approach based on an approximated l_0 (AL0) norm at the receiving terminal, to search the sparse solution via the filtered steepest descent method and projections. The sparse estimation is used for reconstruction of collected data, combing with the previously used measurement matrix and dictionary-matrix. Experimental results confirm the superior performances of the strategies of the proposed method than the traditional methods including matching pursuit (MP) and orthogonal matching pursuit (OMP) methods.

3pSPa3. Experimental study of efficient multi-band underwater acoustic communication algorithm. Hui Su Lee, Chang Uk Baek, Jung-Hyun Seo, Ji Won Jung (Radio Commun. and Eng., Korea Maritime and Ocean Univ., 430,College of Engineering1, 727, Taejong-ro, Yeongdo-gu, Busan 49112, South Korea, lhs6778@kmou.ac.kr), and Dae Won Do (Agency for Defense Development, Changwon, South Korea)

In the underwater acoustic communication environment, multipath transfer characteristics or Doppler spread due to time and space changes such as seabed, sea level, and water depth affect the performance. The multi-band communication technique is effective in terms of performance and throughput efficiency because it can overcome selective frequency fading by allocating the same data to different frequency bands in the environment of rapidly changing channel transfer characteristic. In addition, the transmission distance can be further extended while overcoming various underwater channel environments. However, the multi-band configuration may have worse performance than the single-band one. It is because the performance degradation in a particular band affects the output from the entire bands, which is input into a decoder, thereby decreasing the overall performance. This problem can be solved through a receiving end that analyzes error rates of each band, sets threshold values, and allocates lower weights to inferior bands. In this paper, we proposed an algorithm to set the threshold value using the preamble error rate, which is known data to be transmitted and received. We have analyzed the efficiency of multi-band transmission scheme by applying 1–4 number of multi-bands using turbo pi code through actual underwater experiment.

3pSPa4. Study on the structure of an efficient receiver for multi-sensors using direct sequence spread spectrum. Ahyun Lee, Chang Uk Baek, Ji Won Jung (Radio Commun. and Eng., Korea Maritime and Ocean Univ., Yeongdo-gu, Busan, Busan 13557, South Korea, ahean@kmou.ac.kr), and Dae Won Do (Agency for Defense Development, Changwon, South Korea)

In underwater acoustic communication, there has been a lot of interest in detection and acquisition of information about multiple sensors as well as detection of information on a single sensor. However, research on multiple sensors access is lacking. In order to acquire information from each sensor in multiple sensors access underwater communication, we mainly use the spreading method which can obtain information of each sensor individually at the receiving side by multiplying the code with different orthogonal components in the same frequency band. In the case of multiple sensors, successive interference cancellation is used to remove interference from other sensor and apply the RAKE method. However, in a multiple sensor communication environment, if the channel information is not perfect, the performance is reduced. In this paper, we propose a transceiver structure for multiple sensor covert acoustic communication using the spread spectrum method and RAKE method, and propose and effective receiver structure to overcome drawbacks. An actual underwater experiment was conducted to analyze the performance of the covert underwater acoustic communication. The result of experiment indicated that the transmission and receive structure proposed through the actual underwater experiment was well-suited to the communication model for covert multiple accesses.

3pSPa5. Ultrasonic through-tissue communication with video capabilities. Gizem Tabak, Michael Oelze (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 119 Coordinated Sci. Lab., 1308 W Main St., Urbana, IL 61801, tabak2@illinois.edu), and Andrew Singer (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

The use of wireless implanted medical devices (IMDs), which communicate data wirelessly from sensors within the body to a receiver outside of the body, are gaining significant momentum in medical diagnosis and treatment procedures. Currently, radio frequency (RF) electromagnetic waves are the most frequently used communication method for wireless IMDs. However, there are various drawbacks of using RF electromagnetic waves in such applications such as high attenuation in the body and strictly regulated frequency spectrum, which consequently limit data rates of these devices. An alternative approach to RF transmission for IMDs uses ultrasonic waves, which experience lower attenuation in the body and have higher available bandwidth. In our work, we aim to demonstrate high data rate ($>$1.2 Mbps) through tissue communication using ultrasonic waves that will allow us to stream video with an IMD while actively controlling it inside the body. Initial experiments performed with 2mm biocompatible sonomicrometry transducers, where 2.4 Mbps data rate BER less than 5E-5 through water is achieved, demonstrating the capability of this method for high speed data transmission. Additional targets of this work include data transmission through beef liver and a live rabbit abdominal wall with high enough data rates capable of video streaming.

3pSPa6. Nonlinear waveform distortion: Assessment and detection of clipping on speech data and systems. John H. L. Hansen, Allen Stauffer, and Wei Xia (Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, john.hansen@utdallas.edu)

Speech, speaker, and language systems have traditionally relied on carefully collected speech material for training acoustic models. There is an overwhelming abundance of publicly accessible audio material available for training. A major challenge, however, is that such found data is not professionally recorded, and therefore may contain a wide diversity of background noise, nonlinear distortions, or other unknown environmental or technology-based contamination or mismatch. There is a critical need for automatic analysis to screen such unknown data sets before acoustic model development training, or to perform input audio purity screening prior to classification. In this study, we propose a waveform based clipping detection algorithm for naturalistic audio streams and analyze the impact of clipping at different severities on speech quality measures and automatic speaker identification systems. We use the TIMIT and NIST-SRE08 speech corpora as case studies. The results show, as expected, that clipping introduces a nonlinear distortion into clean speech data, which reduces speech quality and performance for speaker recognition. We also investigate what degree of clipping can be present to sustain effective speech system performance. The proposed detection system, which will be released, could potentially contribute to new audio collections for speech and language technology development.

3pSPa7. Numerical analysis for source direction finding using a cylindrical array with scattered sound fields. Sea-Moon Kim (Korea Res. Inst. of Ships and Ocean Eng., 32 Yuseong-daero 1312beon-gil, Yuseong-gu, Daejeon 34103, South Korea, smkim@kriso.re.kr), Keunhwa Lee (Dept. of Defense Systems Eng., Sejong Univ., Seoul, South Korea), and Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea)

Numerous studies on beamforming techniques have been done for source direction finding using various types of arrays. A circular array as well as a linear array is also widely used for its compact configuration and smaller angular dependency of beamforming performance. In underwater environments, a circular array faces high drag forces due to not only ocean current but also its moving platform. One of the solutions may be an array of hydrophones flush mounted on a cylindrical structure to reduce the drag. However this configuration causes sound scattering, which deteriorate the pressure distribution based on the free field condition. This talk deals with numerical approach for source localization techniques with scattered sound fields. Parametric study shows that beamforming performance is improved and relative high resolution can be achieved using an array of small-sized aperture. [This work was financially supported by the research project PES9410 funded by KRISO.]

3pSPa8. High-resolution ultrasound camera with sparse array for machine diagnosis. Choon-Su Park (Ctr. for Safety Measurements, Korea Res. Inst. of Standards and Sci., Gajeong-ro 267, Bldg. # 206 / Rm. # 206, Daejeon 34113, South Korea, choonsu.park@kriss.re.kr)

Fault detection plays a crucial role in maintenance of mechanical systems. Ultrasound is known to be a prevalent feature of incipient fault. Industrial ultrasound detectors are widely used to find out initial malfunction of machinery. Especially, ultrasound camera can shows the fault by overlapping beamforming power distribution with optical camera image. One can easily observe where the faults are. Ultrasound transducer usually has larger size than the wavelength of its resonance frequency for highly sensitive measurement. It must be a useful approach to design sparse array that gets over the spatial aliasing problem. The aliased beamforming distribution contains high grating-lobe level due to rough spatial sampling beyond half wavenumber spacing. The sparse array is optimized to minimize maximum side-lobe level by genetic algorithm under given constraints. Being with the constraints of size and number of elements, a limitation on spatial resolution is inevitable. Functional beamforming is simple but effective to dramatically increase spatial resolution as well as reduce side-lobes. A high resolution imaging scheme based on functional beamforming with an optimal sparse array is introduced.

3pSPa9. Acoustic and ultrasound investigation of word-initial gemination in Moroccan Arabic. Mohamed Yassine Frej (The MARCS Inst., Western Sydney Univ., 2 Bullebourt Ave., Milperra, NSW 2214, Australia, y.frej@westernsydney.edu.au), Christopher Caringnan (Inst. of Phonet. and Speech Processing (Ludwig-Maximilians-Universität München), Munich, Germany), Michael I. Proctor (Dept. of Linguist, ARC Ctr. of Excellence in Cognition and Its Disord., Macquarie Univ., NSW, Australia), and Catherine T Best (The MARCS Inst., Western Sydney Univ., Milperra, NSW, Australia)

Moroccan Arabic uses geminate/singleton contrasts in medial position but it is controversial whether it maintains them utterance-initially. To address this issue, we made simultaneous ultrasound and acoustic recordings of five native speakers producing target words containing /t/-/tt/ and /d/-/dd/ contrasts utterance-initially and -medially, 10 times each. The ultrasound data were analysed via a novel method of using pixel-derived principal components and linear discrimination to generate a time-varying articulatory "closure" signal directly from the ultrasound images, without the need for manual tracing. This articulatory signal was subsequently used to measure closure duration by determining gestural landmarks from its velocity function. The results provide clear *articulatory* evidence that speakers produce utterance-initial singletons versus geminates with significantly different closure durations, although the contrast provides no *acoustic* evidence of closure duration per se. Rather, our results reveal that speakers realize geminate/singleton contrasts via acoustic dimensions not related to consonant duration: The vowel is significantly longer following geminates than singletons, and the stop bursts have greater amplitude for geminates than singletons. Together these findings provide evidence that the gemination contrast is maintained in initial position in Moroccan Arabic, and challenge traditional assumptions that consonant length contrasts are primarily carried by acoustic closure duration differences.

3pSPa10. SoilComm: A miniaturized through-soil wireless data transmission system. Sijung Yang (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1308 West Main St., Coordinated Sci. Lab., Urbana, IL 61801, Syang103@illinois.edu), Omar Baltaji, Youssef M. Hashash (Civil and Environ. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Andrew Singer (Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Champaign, IL)

Wireless underground sensor networks (WUSN), are emerging Internet-of-things (IoT) technologies, that can benefit numerous applications including geotechnical data acquisition for online infrastructural health monitoring and automated agricultural systems. State-of-the-art approaches to wireless underground sensor communications have employed, as their data carrier, electromagnetic waves, characterized by: their extreme signal path losses in soil due to absorption and scattering, the challenges they impose on transceiver miniaturization, and the power/energy efficiency constraints they suffer from during battery-based operation. Acoustic waves can resolve many of these challenges, benefiting from lower levels of absorption and scattering losses at target frequencies, and from looser size limitations, resulting in smaller form factor, power-efficient transceivers. In this work, biologically inspired, and afterwards motivated by the physical properties of acoustic propagation in soil, a scheme for wireless data communication through soil employing acoustic waves is presented. Experimental results illustrate the system capability of sending application-specific data, ranging from sensor readings to low-resolution images, over distances exceeding 30 m through soil.

3pSPa11. Measuring speech perception with recovered envelope cues using the peripheral auditory model. Nursadul Mamun (EE, Univ. of Texas, Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, nursad49@gmail.com), Khadija Akter (ETE, Chittagong Univ. of Eng. and Technol., Chittagong, Bangladesh), Hussnain Ali, and John H. L. Hansen (EE, Univ. of Texas, Dallas, Richardson, TX)

Speech perception refers to how understandable speech produced by a speaker would be by a listener. The human auditory system usually interprets this information using both envelope (ENV) and temporal fine structure (TFS) cues. While ENV is sufficient for understanding speech in quiet, TFS cues are necessary for speech segregation in noisy conditions. In general, ENV can be recovered from the TFS (known as recovered ENV); however, the degree of ENV recovery and its significance on speech perception are not clearly known/understood. In order to systematically assess the relative contribution of the recovered ENV for speech perception, this study proposes a new speech perception metric. The proposed metric employs a phenomenological model of the auditory periphery developed by Zilany and colleagues (J. Acoust. Soc. Am. 126, 283–286, 2014) to simulate the responses of the auditory nerve fibers to both original and recovered ENV cues. The performance of the proposed metric was evaluated under different types of noise (both steady-state and fluctuating noise), as well as several classes of distortion (e.g., peak-clipping, center-clipping, and phase jitter). Finally, to validate the proposed metric, the predicted scores were compared with subjective evaluation scores from behavioral studies. The proposed metric indicates a statistically significant correlation for all cases and accounts for a wider dynamic range compared to the existing metrics.

3pSPa12. Bat-inspired dynamic features and factors that modulate their impact on speech recognition. Alexander Hsu, Jin-Ping Han, Xiaodong Cui, Kartik Audhkhasi (IBM T. J. Watson Res. Ctr., 1101 Kitchawan Rd., Yorktown Heights, NY, Alexander.Hsu@ibm.com), Anupam Kumar Gupta (Virginia Tech, Blacksburg, VA), Joseph Sutlive (Virginia Tech, Roanoke, VA), Tabassum Ahmed, and Rolf Müller (Virginia Tech, Blacksburg, VA)

One of the most serious remaining challenges in speech recognition is dealing with corruption of speech signal by other nuisance speech ("babble"). A promising approach to solving the problem of separating the signal of interest from the detractors is to inject direction dependent signatures into all signals received, which has been realized by bat-inspired biomimetic pinna—dynamic periphery. Changing the shape of a biomimetic pinna during the recordings introduces substantial time-variant signatures into speech signals. To investigate the utility of these signatures, we have used bioinspired signal representations (cochleagram and spikegram) as input for speech classifiers based on Gaussian mixture models (GMM) and hidden Markov models (HMM). The speech samples used were obtained from open source databases: spoken digits and alphabets from Carnegie Mellon University were mixed with babble or noise samples from Columbia University. Since the time-variant signatures were found to depend strongly on the direction of the sound source, we attempted to include datasets from different directions for training and testing to feed into the classifiers. The results indicate that dynamic periphery can substantially improve recognition and that these effects depend on the signal representation as well as the angular composition of the training dataset.

3pSPa13. Is explicit formant encoding useful for speech perception with cochlear implants? Juliana N. Saba (Univ. of Texas at Dallas, 800 W Campbell Rd., Pflugerville, TX 75080, juliana.saba@utdallas.edu), Hussnain Ali, Colin Brochtrup, and John H. L. Hansen (Univ. of Texas at Dallas, Richardson, TX)

Earlier generations of cochlear implant (CI) sound processing strategies incorporated explicit encoding of formant frequencies. Many of these CI strategies have yielded higher intelligibility outcomes for CI users due to the pure spectral-based approaches used in contemporary processor today, such as: CIS, ACE, Hi-Res, etc. The aim of the study has been to assess intelligibility due to possible loss in formant frequency encoding and its influence on channel selection in diverse acoustic conditions. Formant accuracy from four computer-aided estimation techniques are compared against hand-marked frequencies derived from phoneme-level transcription with sentences from the IEEE and AzBio databases in three noise types at different SNRs: (a) babble, (b) speech-shaped, and (c) noise and reverb combination. CI speech intelligibility outcomes of the proposed strategy using explicit formant frequency encoding was compared to ACE, the standard energy-based '*n*-of-*m*' approach. A relationship between the estimation techniques and differences in channel selection was systematically assessed. Results suggest that minor loss of formant precision does not yield statistically significant differences in resulting speech intelligibility. However, availability of accurate formant cues may help with intelligibility gains in challenging noise such as babble.

3pSPa14. The CCi-MOBILE Vocoder. Hussnain Ali (Elec. Eng., The Univ. of Texas at Dallas, 800 W Campbell Rd., EC33, Richardson, TX 75080, hussnain@ieee.org), Nursadul Mamun (Elec. Eng., The Univ. of Texas at Dallas, Dallas, TX), Avamarie Bruggeman, Ram Charan M. Chandra Shekar (Elec. Eng., The Univ. of Texas at Dallas, Richardson, TX), Juliana N. Saba (Elec. Eng., The Univ. of Texas at Dallas, Pflugerville, TX), and John H. L. Hansen (Elec. Eng., The Univ. of Texas at Dallas, Richardson, TX)

The UT-Dallas Costakis Cochlear Implant Mobile (CCi-MOBILE) research platform enables experimental research with hearing-aid (HA) and cochlear-implant (CI) devices. The platform substitutes the clinical sound processor (CI/HA) with an Android smartphone/tablet or a PC as a computing hardware to run custom speech processing algorithms. The flexibility offered by such a setup enables researchers to provide custom-designed electric and acoustic stimuli (EAS) to CIs and HAs bilaterally in a time-synchronized manner. With this bimodal (electric + acoustic) capability, the platform can be used to undertake studies with individuals who either use a CI in one ear and a HA in the contralateral ear, or use hybrid implants in one or both ears. The CCi-MOBILE software suite includes a range of research applications which address different experimental needs. In the present work, a real-time VOCODER is incorporated in the CCi-MOBILE platform to facilitate research involving acoustic simulations of CIs. Two traditional flavors of VOCODER processing are implemented, namely noise-band and sine-wave vocoding. Flexibility is provided to modify processing parameters such as number of channels, frequency band-widths, filter attributes, etc. This flexibility combined with the ability to conduct long-term studies beyond the laboratory in diverse naturalistic acoustic environments will help advance research in psychoacoustics.

Session 3pSPb

Signal Processing in Acoustics: Geometric Signal Processing in Acoustics

Ananya Sen Gupta, Cochair
Electrical and Computer Engineering, University of Iowa, 4016 Seamans Center for the Engineering Arts and Sciences, Iowa City, IA 52242

Jeffrey S. Rogers, Cochair
Acoustics Division, Naval Research Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375

Chair's Introduction—1:00

Invited Papers

1:05

3pSPb1. Implications of Riemannian geometry in underwater acoustics. Steven I. Finette (Acoust. Div., Code 7160, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

Signal processing involving source localization and detection in underwater acoustics often depends on comparing covariance or cross-spectral density matrices (CSDMs) estimated for data, replica or noise source data from passively sensed acoustic fields incident on sensor arrays. Such comparisons often depend on a measure of similarity between matrix pairs and this typically involves a Euclidean metric. While Euclidean-based metrics are ubiquitous, convenient and useful, they are not fundamental to this task. By exploiting the facts that a CSDM is Hermitian and positive semi-definite, one can interpret such matrices as points constrained to a Riemannian, rather than a Euclidean manifold, with the implication that similarity between matrix pairs should be measured using a metric consistent with the manifold's intrinsic structure. This geometric interpretation leads to alternative matched-field processors for source localization involving the Riemannian distance as a measure of similarity and can be used to solve this inverse problem. Some implications of this non-Euclidean approach in underwater acoustics, as well as a possible extension to source detection are discussed. [Work supported by the Office of Naval Research.]

1:25

3pSPb2. The Biquaternionic acoustic wave equation. Roger M. Oba (Acoust. Div., Code 7167, Naval Res. Lab., 4555 Overlook Ave. SW, Acoust. Div., Code 7167, Washington, DC 20375, roger.oba@nrl.navy.mil)

Equations for irrotational, linearized Euler fluid, where the mass continuity equation incorporates a linear equation of state, define linear acoustics that can be recast in terms of complexified quaternions, the biquaternions, which satisfy Cauchy-Fueter regularity equations. This gives first order partial differential equations from the four space-time dimensions to the four (complex) dimensional velocity-pressure space, with time and pressure in the biquaternionic scalar parts. The biquaternionic planar exponential provides the basis to elementary solutions as biquaternionic functions of a single biquaternionic variable. Analysis of this form shows significant geometrical algebraic properties including imaginary units for time distinct from those for space. The single biquaternion acoustic field at each point of space-time allows the propagation four complex coefficients in a single geometric algebraic structure, suitable for three or four dimensional Fourier analysis. (Work supported by the U.S. Office of Naval Research.)

1:45

3pSPb3. Fermat's principle, Fresnel zones, and ray methods for high-frequency signatures of elastic cylinders in water. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Geometric methods are useful for understanding the magnitude and timing of high-frequency echoes from elastic objects in water. This presentation emphasizes, but is not limited to, elastic contributions associated with leaky waves on circular cylinders. The regions of strong coupling between the incident acoustic wave and guided leaky waves can be predicted using Fermat's principle while the size of the associated Fresnel zones are useful for understanding signal magnitudes [P. L. Marston, J. Acoust. Soc. Am. 102, 1628–1638 (1997); F. J. Blonigen and P. L. Marston, 110, 1764–1769 (2001)]. Ray theory predictions are supported by measurements with tilted cylinders [S. F. Morse, P. L. Marston, and G. Kaduchak, J. Acoust. Soc. Am. 103, 785–794 (1998); K. Gipson and P. L. Marston, J. Acoust. Soc. Am. 106, 1673–1680 (1999); F. J. Blonigen and P. L. Marston, J. Acoust. Soc. Am. 112, 528–536 (2002)]. Related applications include signatures and images of tilted cylinders near flat surfaces [D. S. Plotnick and P. L. Marston, J. Acoust. Soc. Am. 140, 1525–1536 (2016)], time-frequency signatures [S. F. Morse and P. L. Marston, J. Acoust. Soc. Am. 111, 1289–1294 (2002)], and signatures for other targets with corners. [Work supported by ONR.]

2:05

3pSPb4. Adaptive integration for Capon-type beamformers by exploring Riemannian geometry on covariance matrices. Magnus L. Nordenvaad (Marine Systems, Swedish Defence Res. Agency (FOI), Gullfossgatan 6, Stockholm, Sweden 16490, Sweden, maglun@foi.se)

Adaptive, or Capon beamformers, offers several advantages compared to standard alternatives in underwater sensing. Furthermore, for hull mounted sensor arrays, adaptive beamformers is a convenient way of mitigating platform induced interference. The underlying difficulty in implementing a data-adaptive approach is how to obtain an accurate estimate of the data covariance matrix, R. The combination of high-dimensionality and time-variation renders significant challenges in maintaining accurate covariance estimates. A common way to approach this issue is to update the covariance matrix using a weighted Euclidean average, R_{new} = a R_{old} + (1-a) R_{inst}, where R_{inst} is some instantaneous measurement. Meanwhile, the time-variation naturally form a trajectory on the covariance matrix manifold. With this in mind, it makes much more sense to smooth the trajectory on the space of covariance matrices using proper metrics. The purpose of this paper is to take the geometry of covariance matrices into account while performing adequate covariance matrix estimation. Specifically, we propose a simple and elegant algorithm to track the covariance based on an update through means on Riemannian manifolds. This solution is then incorporated into an adaptive beamformer, and initial evaluations based on both simulated and real data show slight improvements compared to standard approaches.

Contributed Papers

2:25

3pSPb5. Fast two-dimensional censored mean level detector-statistics constant false alarm rate method for multi-beam seafloor terrain detection. Jiaqi Wang and Haisen Li (Harbin Eng. University, China, 145th, Str. Hantong, Harbin, Heilongjiang Province 150001, China, wangjiaqi@hrbeu.edu.cn)

In order to solve the problem of false terrain caused by environmental interference and "tunneling effect" in conventional multi-beam seafloor terrain detection, this paper addresses a seafloor topography detection method based on fast two-dimensional (2D) CMLD-CFAR (censored mean level detector-Statistics Constant False Alarm Rate). A cross-sliding window is used in this method. The target occlusion phenomenon which occurs in multi-target environments can be eliminated by censoring the large cells of the reference cells, and the rest reference cells are used to calculate the local threshold. The conventional 2D CMLD-CFAR methods need to estimate the background clutter power level for every pixel, which costs a lot of computational load. This paper proposes a fast algorithm with a global threshold to reduce the computational load. Only the regions of interest (ROI) which are selected by a global threshold will be detected by 2D CMLD-CFAR method, and the rest pixels are distinguished as clutters directly. The proposed method has been evaluated by the real multi-beam experimental data of which the background follows a distribution of Exponential. The results show that the novel method can effectively solve the problem of false terrain in multi-beam terrain survey and has a high detection accuracy.

2:40

3pSPb6. Representing sonar target features using Braid geometry. Ananya Sen Gupta (Elec. and Comput. Eng., Univ. of Iowa, 4016 Seamans Ctr. for the Eng. Arts and Sci., Iowa City, IA 52242, ananya-sengupta@uiowa.edu) and Ivars P. Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

Acoustic color features that specify a sonar target in the time-frequency spectral domain may be broadly classified into two groups: (i) elastic wave features that depend on the material composition of the target, and (ii) wave features that depend on the specific geometry of the sonar target. Typically these two types of target features overlap intricately in the acoustic color domain that renders them difficult to separate using traditional signal processing techniques. Furthermore, field experimental conditions such as target orientation, proud, buried or semi-buried state of the target as well as sediment characteristics, influence the relative overlap and morphology of these acoustic color features. A key bottleneck to robust target feature representation and subsequent classification is how these potentially overlapping features may change in the field from what is measured under laboratory conditions or observed through model-based simulations. We will present some relevant results from our ongoing work in disentangling sonar features using a combination of geometric signal processing, graph-based informatics techniques and well-known physical models. Specifically, we will employ geometry of braid manifolds as a possible basis for representing the physical phenomena influencing each group of target features. We will also present recent results from graph-based informatics to show how braid encoding across multiple experimental conditions can be employed to determine and disentangle overlapped target features.

2:55

3pSPb7. Harnessing geometric techniques for robust real-time estimation of shallow water acoustic channels. Ananya Sen Gupta and Ryan McCarthy (Elec. and Comput. Eng., Univ. of Iowa, 4016 Seamans Ctr. for the Eng. Arts and Sci., Iowa City, IA 52242, ananya-sengupta@uiowa.edu)

The shallow water acoustic channel is well-known to exhibit rapid fluctuations in delay spread, which poses a fundamental roadblock to real-time channel estimation, particularly under moderate to rough sea conditions. This talk will summarize recent research on multiple representations of the shallow water acoustic channel and present the relative merits of each channel representation in localizing different channel effects. In particular, we will examine the efficacy of employing geometric signal processing techniques to adaptively track the shallow water acoustic channel in each representation. Results based on experimental field data as well as channel simulations under varying environmental conditions will be presented.

3:10–3:30 Panel Discussion

ASA Plenary Session and Awards Ceremony/CAA Annual General Meeting

Lily M. Wang, Cochair
President, Acoustical Society of America

Jérémie Voix, Cochair
President, Canadian Acoustical Association

Traditional Lkwungen Territory Welcome

Acoustical Society of America Plenary Session and Awards Ceremony

3:45 p.m. to 4:45 p.m.

Presentation of ASA Fellowship Certificates

Ahmed A. Al-Jumaily – For contributions to biomedical applications of acoustics and vibrations

Richard D. Costley, Jr. – For contributions to the analysis of battlefield acoustics and the acoustical properties of structures

Jan Dettmer – For contributions to inversion methods and uncertainty quantification applied to geoacoustics

Joseph R. Gladden, III – For service to and leadership in the field of physical acoustics

Dorian S. Houser – For advancing the understanding of the impacts of anthropogenic noise on marine mammals

Brian F. Katz – For contributions to measure teechniques and spatial hearing for auditory virtual reality

Colleen Reichmuth – For contributions to pinniped acoustics

Gary P. Scavone – For contributions to the analysis and modeling of musical instruments

Eleanor P. Stride – For contributions to the modelling, development, and manufacturing of acoustically responsive biomaterials

David S. Woolworth – For contributions to general architectural acoustics and service to the Society

Catherine L. Rogers – For contributions to speech communication through service, mentoring, and scholarship

Presentation of Acoustical Society Awards

Wallace Clement Sabine Medal to Michael Vorländer

Canadian Acoustical Association Annual General Meeting

4:45 p.m. to 5:30 p.m.

Presentation of Canadian Acoustical Association Awards

Shaw Postdoctoral Prize in Acoustics to Olivier Valentin (École de technologie supérieure, Montréal) for "EARtrode, A Wireless In-ear Custom-fitted intelligent Brain Computer Interface"

Bell Student Prize in Speech Communication and Hearing to Megan Keough (University of British Columbia, Vancouver) for "Reafferent Feedback and Aerotactile Integration"

Fessenden Student Prize in Underwater Acoustics to Josee Belcourt (University of Victoria) for "Bayesian Geoacoustic Inversion of Seabed Reflection Data at the New England Mud Patch"

Bregman Student Prize in Psychological Acoustics to Sean Gilmore (Ryerson University, Toronto) for "Feeling the Beat: An Investigation into Tactile Beat Perception"

Northwood Student Prize in Architectural and Room Acousticsto Magdalenn Bahour (Ryerson University, Toronto) for "Living Wall and Acoustic Comfort - A Case Study"

Canada-Wide Science Fair Award to Zachary Trefler (Waterloo Collegiate Institute) for "VoiceShield: Teaching Computers to Distinguish Real Data From Fake"

Session 3eED

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

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

Tracianne B. Neilsen, Cochair
Brigham Young University, N311 ESC, Provo, UT 84602

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

This workshop for Victoria Girl Guides consists of hand-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.

OPEN MEETINGS OF TECHNICAL COMMITTEES/SPECIALTY GROUPS

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday. See the list below for the exact schedule.

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

Committees meeting on Tuesday, 6 November

Committee	Start Time	Room
Engineering Acoustics	4:30 p.m.	Rattenbury A/B (FE)
Acoustical Oceanography	7:30 p.m.	Esquimalt (VCC)
Animal Bioacoustics	7:30 p.m.	Oak Bay 1/2 (VCC)
Architectural Acoustics	7:30 p.m.	Theater (VCC)
Physical Acoustics	7:30 p.m.	Colwood 1/2 (VCC)
Psychological and Physiological Acoustics	7:30 p.m.	Salon B (VCC)
Speech Communication	7:30 p.m.	Salon A (VCC)
Structural Acoustics and Vibration	8:00 p.m.	Saanich 1/2 (VCC)

Committees meeting on Wednesday, 7 November

Committee	Start Time	Room
Biomedical Acoustics	7:30 p.m.	Sidney
Signal Processing in Acoustics	7:30 p.m.	Colwood 1/2

Committees meeting on Thursday, 8 November

Committee	Start Time	Room
Computational Acoustics	4:30 p.m.	Esquimalt (VCC)
Musical Acoustics	7:30 p.m.	Crystal Ballroom (FE)
Noise	7:30 p.m.	Shaughnessy (FE)
Underwater Acoustics	7:30 p.m.	Rattenbury A/B (FE)

WALLACE CLEMENT SABINE AWARD OF THE ACOUSTICAL SOCIETY OF AMERICA

Michael Vorländer

2018

The Wallace Clement Sabine Award is presented to an individual of any nationality who has furthered the knowledge of architectural acoustics, as evidenced by contributions to professional journals and periodicals or by other accomplishments in the field of architectural acoustics.

PREVIOUS RECIPIENTS

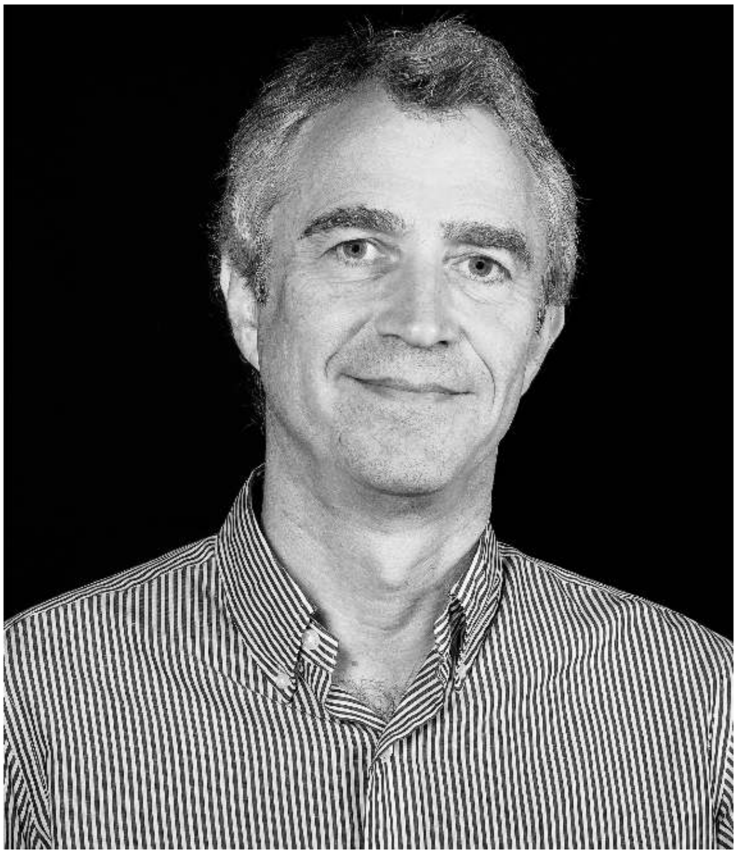
Vern O. Knudsen	1957
Floyd R. Watson	1959
Leo L. Beranek	1961
Erwin Meyer	1964
Hale J. Sabine	1968
Lothar W. Cremer	1974
Cyril M. Harris	1979
Thomas D. Northwood	1982
Richard V. Waterhouse	1990
A. Harold Marshall	1995
Russell Johnson	1997
Alfred C. C. Warnock	2002
William J. Cavanaugh	2006
John S. Bradley	2008
J. Christopher Jaffe	2011
Ning Xiang	2014
David Griesinger	2017

SILVER MEDAL IN ARCHITECTURAL ACOUSTICS

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 RECIPIENT

Theodore J. Schultz 1976



CITATION FOR MICHAEL VORLÄNDER

. . . for contributions to room-acoustic simulations and virtual auditory displays

7 NOVEMBER 2018 • VICTORIA, CANADA

Michael Vorländer was born in Duisburg, Germany. At age 17 he was interested in basketball and playing jazz but during his high school years he became intrigued by physics, mathematics, and astronomy. Also, during this time Michael met his wife Angelika while both were still in high school. He began his university studies in physics at the University of Dortmund in 1978. During that period he was accepted as a student intern at the Institut für Bauphysik (IfB) which led him to think of combining his own interests with those of his father's in physics and architecture. For the IfB Michael conducted field measurements on sound insulation and programmed protocols on a HP 41C PRM calculator. This internship was the origin of his journey into acoustics.

In 1980, Angelika was accepted to study Art and Design in Aachen, which led Michael to transfer to the RWTH Aachen University after his 'Vordiplom' (prerequisite for a full Diploma). At the RWTH Aachen, Michael took solid-state physics and astronomy classes, along with courses in acoustics taught by Professor Heinrich Kuttruff. In his Master's thesis research, Michael calculated Michael Barron's lateral energy ratio with image sources in performance venues on an Apple II. This research resulted in his first peer-reviewed journal paper co-authored with Professor Kuttruff in ACUSTICA 58, 118-129, 1985.

Michael and Angelika married in 1984 and he began his doctoral research with Professor Kuttruff in 1985. He completed his doctoral dissertation in 1989 with three additional journal publications. Among them was his proposal for a hybrid ray-tracing and image source method which led to great progress in efficient computer simulations in room-acoustics (Journal of the Acoustical Society of America 86, 172-178, 1989).

Following his doctoral degree, Dr. Vorländer accepted a postdoctoral position at the PTB Braunschweig, the National Institute of Metrology in Germany. Here Michael shifted his research focus to audiological measurements, binaural technology and microphone calibration. Two years later, in 1991, Michael was appointed the Head of the laboratory of building acoustics at the PTB, which was responsible for room and building acoustics, sound emission and absorption measurement along with microphone calibration in reverberation chambers.

While working for the PTB, Dr. Vorländer completed the 'habilitation' (the inauguration dissertation) in 1994 at the Technical University of Dresden. This led Michael to be named a Professor at the RWTH Aachen in 1996, succeeding Professor Kuttruff, as the head of the Institute of Technical Acoustics (ITA). Professor Vorländer expanded his research to a wide variety of high-impact topics including room-acoustics and virtual auditory reality. He leads an extremely productive research team that, so far, has amassed over 100 peer-reviewed journal articles.

Michael has authored a monograph entitled *Auralization* (Springer, 2007), which has become an influential volume in the architectural acoustics field and the virtual auditory reality community, and has been cited over 540 times. He has also contributed to 15 book chapters including his latest chapter 'Auralization' for the *Architectural Acoustics Handbook* (J. Ross Publishing 2017, Ed. N. Xiang). In another milestone contribution to architectural acoustics, Michael and Dr. E. Mommertz, defined the scattering coefficient of interior surfaces in enclosures along with a methodology to experimentally measure it (Journal of Applied Acoustics 60, 187-199, 2000). This contribution made the materials and devices of architectural acoustics rigorously quantifiable, which considerably advanced the room-acoustics and computer-simulation field. Since then, this work was also adopted as an International Standard (ISO 17497-1, 2004).

Dr. Vorländer is a Fellow of the Acoustical Society of America (ASA). He has served as a member and chair of the Society's Committee on International Research and Education (CIRE) and he was recently elected a Member of the Executive Council. Michael has served as a JASA Associate Editor for Architectural Acoustics since 2013. He has presented many invited papers as well as organized or co-organized a number of architectural acoustics sessions at ASA meetings.

Dr. Vorländer has played significant roles at the scientific and administrative leadership levels in German, European, and international communities. He served as Vice President and President of the German Acoustical Society (2013-2016, 2016-2019), Vice President and President of the European Acoustics Association (2004-2007, 2007-2010), President of the International Commission for Acoustics (2010-2013), and Editor-in-Chief of the journal Acta Acustica united with Acustica (1998-2003). His international meeting service includes Technical Co-chair of the International Congress on Acoustics (ICA) in 2007, General Co-chair of Acoustics '08 Paris, and General Chair of the ICA congress to be held in 2019 in Aachen. He was awarded the R. W. B. Stephens Medal of the Institute of Acoustics in 2005 and the 2014 EAA Award for contributions to the promotion of acoustics in Europe.

Recreationally, he enjoys playing drums and is often seen at the Jam Sessions at ASA meetings. Even though there is no basketball for Michael these days, he still plays soccer with his ITA colleagues.

There is no doubt that Dr. Michael Vorländer has made distinguished contributions to acoustics research, publications, and leadership in acoustical societies. We are delighted that the Acoustical Society of America recognizes and honors Dr. Michael Vorländer with this prestigious Wallace Clement Sabine Medal.

NING XIANG
LAURI SAVIOJA