Session 4aAA

Architectural Acoustics: Emerging Parametric/Generative Design Tools in Architectural Acoustics

Ian B. Hoffman, Chair
Dept. of Architecture, Judson Univ., 1151 N State Street, Elgin, IL 60123

Chair's Introduction—8:30

Invited Papers

8:35

4aAA1. Generative acoustics in architecture. Emily Schilb (Edward Dugger + Assoc., P.A., Stuart, VA), Michael Ermann (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu), David Rife (Dave and Gabe, Blacksburg, VA), and Andrew Hulva (Architecture + Design, Virginia Tech, Blacksburg, VA)

The researchers share three studies in parametric, generative, and iterative architectural acoustics. (1) A real-time parametric model shapes a geometrically complex ceiling plane with grips in order to visualize first-order reflections so that early arriving strong reflections may be directed to the audience and late-arriving echoes may be avoided. (2) A ray-tracing software model pivots door openings so that a coupled volume concert hall can be calibrated to establish the "sweet spot" range of aperture sizes that are more likely to produce a double-sloped sound decay. (3) A custom-designed auralization software simulates sound transmission loss so that designers may hear the (relative) noise isolation of different assemblies.

9:00

4aAA2. Elucidation of acoustical phenomena through the re-tooling of comprehensive acoustical simulations. Arthur W. van der Harten (Acoust. Distinctions, 145 Huguenot St., New York City, NY 10801, Arthur.vanderharten@gmail.com)

In its 8 years available to the general public, the open source acoustical simulation tool Pachyderm Acoustic for Rhinoceros has been used not only for prediction of room quality and noise level, but in a variety of other ways which help to clarify the physical behavior of sound in space. Since a scripting interface for Ironpython was released in 2011, and a grasshopper interface in 2015, Pachyderm has earned a place in the growing field of customizable workflows. Its geometrical and numerical tools have been re-purposed for customized visualizations, real-time analysis, search algorithms, and even sculpture. This talk exhibits some of the most interesting work done by re-tooling Pachyderm Acoustic to-date, and attempts to speculate on how it may be used in the future, as designers with more knowledge of acoustics begin to emerge in the workplace.

9:25

4aAA3. Parametric design applications in architectural acoustics—Generation and optimization of reflective surfaces for specific source/receiver combinations. Marcus R. Mayell (Threshold Acoust. LLC, 141 West Jackson Blvd., Chicago, IL 60604, mmayell@thresholdacoustics.com) and Ian B. Hoffman (Architecture, Judson Univ., Elgin, IL)

This investigation considers the potentials of what are normatively visual parametric design tools, within acoustics design thinking. The typical workflow inherent to most room acoustic software consists of architectural design followed by analysis. In response to the emergence of parametric tools in architectural design, like Rhino and Grasshopper, acoustic designers now have the potential to carry out some analysis through visualization in a more streamlined manner. Furthermore, parametric tools carry the capability to alter the typical workflow from "design followed by analysis" to a more integrated design and analysis workflow, or in some cases even a set criteria followed by a set of design solutions. Through investigation and application of these idea as a part of a recent Master of Architecture thesis project at Judson University, I have been able to develop and test specific parametric definitions which define and visualize reflecting surface potentials based on set source and receiver area criteria.

9:50

4aAA4. Statistical considerations of early-stage design by using computer-based tools in architectural acoustics. Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de) and Shuai Lu (School of Architecture, Tsinghua Univ., Beijing, China)

Room simulation techniques require input data of the room model and the boundary conditions. In an iterative early-stage design process, the general room shape and the boundary conditions correspond to the perceptual parameter space such as reverberation time, strength, clarity, etc. The influence of specific settings in the algorithmic details such as the number of rays, the temporal resolution, the filter bandwidth will be demonstrated by calculating room impulse responses under the condition of variation of such settings. Furthermore, the statistics of the input variables of the general shape result in certain degrees of freedom for the absorption and scattering coefficients. It is discussed how a design space could be defined, which describes the probability for achieving the desired room acoustic performance as a function of the general room shape and boundary conditions.

10:15–10:30 Break

10:30

4aAA5. Lost in translation, easing communication through the use of digital modeling. Shane J. Kanter, Gregory Miller, and Marcus Mayell (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, skanter@thresholdacoustics.com)

Daily communication between Acoustician, Architect, facility users, building owners, and consultants paves a road riddled with several opportunities for miscommunication. Although many design team members have honed their auditory senses, many of us communicate most effectively through the use of visual graphics and diagrams. Computer aided design, including the use of moving images, has not only become a tool for analysis but also allowed for more effective communication to educate and influence important design decisions. Parametric modeling tools allow consultants to develop analysis and communication tools early on and throughout the design of architectural elements. Two recent applications of the digital modeling process include the new orchestra shell at the Lyric Opera in Chicago and a 1000 Seat Convening Hall. Digital tools used to model and analyze acoustic performance have been imperative while working through the challenges embedded within each project. In both cases, these tools have helped overcome design and communication challenges among the team, leading to a result that meets the unique needs of each space.

10:55

4aAA6. Hope in mirrors: Update on using SketchUp and light rendering to visualize acoustic reflections. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

At a previous ASA meeting, a method was shown that allows architects to visualize acoustics in an intuitive manner using tools already familiar in the design workflow. The method is based upon some of the earliest forms of acoustics modeling, whereby mirror surfaces and light sources were used in scale models to determine the architecture's impact on early reflections (a physical modeling of the image source method, essentially). This talk will demonstrate applications using this revitalized method, as well as expand on the original method using 360-degree immersive imagery, animating discrete reflection orders spatially (as opposed to temporally), and other useful techniques which are afforded by virtual modeling that would never otherwise be permitted by natural physics.

THURSDAY MORNING, 26 MAY 2016 — SALON I, 8:00 A.M. TO 11:00 A.M.

Session 4aAO

Acoustical Oceanography and Animal Bioacoustics: Noise Impacts from the Industrialization of the Outer Continental Shelf and High Seas

Michael A. Stocker, Chair
Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938

Chair's Introduction—8:00

Invited Papers

8:05

4aAO1. Noise sources from the industrialization of the ocean. Michael Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org)

Increasingly technology is opening up hostile and challenging marine environments for industrial exploitation. This is occurring in the energy sector with fossil fuel exploration and extraction operations and developing wind and hydrodynamic energy projects. It is also occurring with the deep-water expansion of other extraction industries such as minerals mining and fishing. All of these operations and enterprises are introducing loud and complex noise sources into marine bioacoustic habitats. This presentation will be an overview examination of existing and developing noise sources that are a consequence of the industrialization of the outer continental shelf and high seas.

8:25

4aAO2. Underwater sound radiation from subsea factories. Bas Binnerts (Acoust. and Sonar, TNO, Oude Waalsdorperweg 64, Den Haag 2597AK, Netherlands, bas.binnerts@tno.nl) and Pieter v. Beek (Fluid Dynam., TNO, Delft, Netherlands)

In the Oil & Gas industry, there is a trend toward more subsea activities such as the processing of the Oil and Gas in so-called "subsea factories." In this work, an overview is presented of the various anthropogenic sources contributing to the soundscape during the operational phase of these factories. With the measured sound spectrum (in air) of a complete turbo compressor installation and the known differences between radiation into air and sea water, a stylized, equivalent monopole source level is constructed. This source level is put into perspective by comparing it against a variety of other anthropogenic continuous sources. For a stylized subsea factory, it was judged that the turbo compressor will dominate the generated sound field of subsea factories. Finally, potential risks of the radiated sound are identified and possible sound mitigation solutions are discussed.

8:45

4aAO3. Underwater sound signatures of offshore industrial operations. Christine Erbe, Kim Allen, Alec Duncan, Alexander Gavrilov, Robert McCauley, Iain Parnum, Miles Parsons, and Chandra Salgado-Kent (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

Marine industries, such as offshore petroleum, minerals, fisheries, transportation, tourism, defence, etc., introduce sound underwater, changing marine soundscapes, ranging from shallow coastal to deeper offshore regions. Concern about potential noise impacts on marine fauna has led to numerous underwater recordings and bioacoustic studies. Prior to operations, e.g., as part of permit applications for marine operations, environmental impact assessments are carried out that rely on the modelling and prediction of sound emission, propagation, and impacts. A catalog of sound signatures from activities ranging from exploration and surveying to construction, production, general operation, and decommissioning is necessary for predictive modeling. Underwater sounds recorded from seismic airguns, subbottom profilers, echosounders and sonars (sidescan, single-beam, and multi-beam), marine traffic (from small boats to large ships), aerial transportation (helicopters recorded underwater), dredging, pile driving, explosions, drilling, floating petroleum production, storage facilities, etc., are reviewed and their spectral and temporal characteristics, as well as beam patterns are discussed.

9:05

4aAO4. Marine soundscape during a shallow-water seismic survey in the Arctic Ocean. Shane Guan (Office of Protected Resources, NOAA/NMFS, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20910, shane.guan@noaa.gov) and Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC)

For noise generating activity that lasts for an extended period of time, an overall increase in noise levels and change of soundscape within a larger area (over tens of km^2) can be expected. This study analyzed the sound field characteristics during a shallow-water marine seismic survey in the Beaufort Sea of the Arctic Ocean. Three bottom mounted acoustic sensors were deployed in the survey area: two outside the barrier islands in water depths about 12 m, and one inside the barrier islands in water depth of 2.8 m. Averaged 1 min sound pressure levels (SPLs) in broadband, 100–500 Hz, 1–5 kHz, and above 10 kHz bands were computed for periods when airguns were active and inactive. The results showed an 8-dB increase during the period when airguns were active in the 100–500 Hz band for two locations outside the barrier islands. However, there was no noticeable difference in SPLs during periods airguns were active and inactive inside the barrier islands. This is probably due to higher natural ambient noise and low-frequency cut-off of airgun pulses in this extreme shallow location.

9:25

4aAO5. Underwater sound directionality of commercial ships. Martin Gassmann, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0205, mgassmann@ucsd.edu)

The underwater sound radiated by commercial ships is an unintended by-product of their operation and one of the most significant contributors to man-made noise at low frequencies in the ocean. To estimate the directionality of underwater sound radiated by current commercial ships, a seafloor array of five high-frequency acoustic recording packages (HARPs) deployed to 1 km depth with a maximum horizontal aperture of 1 km was used. As a ship of opportunity passed over the HARP array, the directions from the ship to each HARP along with the corresponding source levels were estimated for each ship location. Ships were tracked via satellites (Automatic Identification System—AIS) and acoustically by a frequency domain beamformer that was implemented for one of the HARPs configured with a volumetric hydrophone array (2 m maximum aperture). The directionality estimates of contemporary commercial ships exhibit significant stern-bow asymmetries among other quantitative characteristics that will be discussed.

9:45–10:00 Break

10:00

4aAO6. Comparing methods for estimating the injury and behavioral disturbance radii from sound source characterization measurements. Bruce Martin, Jeff MacDonnell (JASCO Appl. Sci., 32 Troop Ave., Ste. 202‘, Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), Alexander O. MacGillivray, and David Hannay (JASCO Appl. Sci., Victoria, Br. Columbia, Canada)

It is a common practice for regulators to require project proponents to estimate the radius around a sound source where marine life could be injured or disturbed. Acoustic propagation modeling is normally required so that realistic radii are obtained that take into account the bathymetry, bottom properties, water column sound velocity profile, as well as source spectrum and directivity. Increasingly proponents are also asked to perform *in-situ* sound source characterization (SSC) measurements to verify the modeling predictions, including the spectrum and directivity of the sound source. During an SSC, sound levels are measured at increasing range from the source. In most cases, it is not possible to measure the sound levels at all ranges of interest, and therefore, the data must be interpolated or extrapolated to estimate the radii of the regulatory sound isopleths. This talk uses real-world data to identify the strengths and weaknesses of four methods of estimating the radii: (1) the practical spreading model; (2) linear interpolation; (3) linear regression with absorption; and (4) model-measured fits.

10:20

4aAO7. The effect of operational measures on shipping sound in the North Sea. Bas Binnerts (Acoust. and Sonar, TNO, Oude Waalsdorperweg 64, Den Haag 2597AK, Netherlands, bas.binnerts@tno.nl) and Christ de Jong (Acoust. and Sonar, TNO, The Hague, Netherlands)

From studies into marine biology, it is known that the behavior of marine mammals and fish can be influenced by the ambient underwater sound level. In busy seas like the North Sea in Europe, sound from shipping traffic is largely responsible for the low frequency part of the ambient sound. In this work, possibilities to regulate the underwater radiated sound of ship traffic are investigated. This involves measures that look at traffic flows and operational use of the vessels, to determine what can be done to reduce the shipping sound levels in certain marine areas. Three types of operational measures are considered: (1) the effect of spatial planning (for example, changing the location of shipping routes), (2) the introduction of a radiated sound limit for vessels in a certain area, and (3) the introduction of a speed limit for vessels in a certain area. The proposed measures have been analyzed on their effect and effectiveness, by means of numerical analysis. Calculations have been performed for a generic ship traffic flow, based on actual recorded AIS data from a North Sea shipping lane, using a speed depended model for the source level of different ship types and a sound propagation model.

10:40–11:00 Panel Discussion

THURSDAY MORNING, 26 MAY 2016 SALON B/C, 8:00 A.M. TO 11:45 A.M.

Session 4aNS

Noise, ASA Committee on Standards, Animal Bioacoustics, Engineering Acoustics, and Physical Acoustics: Wind Turbine Noise I

Nancy S. Timmerman, Cochair
P.E., 25 Upton Street, Boston, MA 02118

Kenneth Kaliski, Cochair
RSG Inc., 55 Railroad Row, White River Junction, VT 05001

Robert D. Hellweg, Cochair
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Paul D. Schomer, Cochair
Schomer and Associates, Inc., 2117 Robert Drive, Champaign, IL 61821

Chair's Introduction—8:00

Invited Papers

8:05

4aNS1. A case study in Canada: Exploring research challenges of industrial wind turbines and health. Carmen M. Krogh (Killaloe, Killaloe, ON K0J 2A0, Canada, carmen.krogh@gmail.com) and Jeffery Aramini (Fergus, ON, Canada)

The topic of adverse health effects associated with industrial wind turbines (IWT) is controversial and debated worldwide. Some residents living in proximity to wind energy facilities report symptoms of sleep disturbance, annoyance, headaches, ear pain/discomfort, mood disorders, stress, cardiac and blood pressure effects, reduced quality of life, and other adverse effects. In some cases, research initiatives have been the result of individuals' complaints. The research is challenged in part by the complexities and numerous variables associated with this subject. A range of IWT research approaches, sometimes in combination with each other, has been used including self-reporting surveys, investigations and acoustical measurements. On July 12, 2012, Health Canada announced its large scale cross-

sectional randomized epidemiological wind turbine noise and health study. The research was conducted by Health Canada in two Canadian provinces. Health Canada indicated its study had limitations, would not be definitive, and would not permit any conclusions to be made with respect to causality. This presentation will explore some of the inherent challenges of studying health effects associated with wind energy facilities and will consider the role of those individuals reporting adverse health effects.

8:25

4aNS2. Industrial wind turbines and adverse health effects: Where we are, where we need to go, and the need for regulations and predictive models to recognize human physiology. Michael A. Nissenbaum (Radiology, McGill Univ., 1 Westmount Square, Ste. C210, Westmount, QC H3Z 2P9, Canada, mnissenbaum@att.net)

Since our initial study was published (Nissenbaum *et al.*, 2012), work in several areas of human physiology has begun to elucidate the precise mechanisms by which sleep disturbances result in adverse health effects, over both short and longer durations. These include impaired neuronal connections in the learning brain, altered genetic expressions impacting the immune system, and correlations between poor quality sleep and MRI-measured atrophy of the brain over a mean period of 3.5 years. Additionally, fMRI has demonstrated brain responses to sounds with frequencies as low as 8 Hz. At lower frequencies, somatosensory mechanisms are now thought to play a role, in addition to auditory. Local regulations regarding noise (Soundscape) limits and methods of measurement were designed prior to current understandings of human sensory and reactive physiology. Instrumentation and modeling geared towards satisfying those regulations are by implication lacking because they do not capture or predict physiological responses to IWT noise. According to the principles of soundscape, and given the subtleties of human physiology, humans remain the best instruments available for detecting objectionable noise and identifying adverse health effects. Regulations, measurement methods, and predictive models must adapt to current understandings of human physiology to best protect human populations.

8:45

4aNS3. Measuring perceptual effects of infrasound. Peggy B. Nelson, Michael Sullivan, Meredith Adams, Matthew Lueker, Jeffrey Marr (Univ. of Minnesota, 164 Pillsbury Dr. Se, Minneapolis, MN 55455, peggynelson@umn.edu), and Bruce Thigpen (Eminent Technologies, Tallahassee, FL)

A multi-disciplinary group of researchers at the University of Minnesota Center for Applied and Translational Sensory Science (cats-s.umn.edu) are designing and pilot testing the perceptual effects of infrasound, in collaboration with Eminent Technologies (rotarywoofer.com). An infrasound generator simulates the acoustic signature of audible sound and infrasound generated by wind turbines in the field. With the support of Xcel Energy, the team of engineers, otologists, hearing scientists, and balance experts are evaluating the effects of infrasound only, acoustic turbine sound, and combined infrasound and acoustic sound. We are testing listeners' quiet detection, masked detection, discrimination, and rating of signals. Pilot results will be described. Together we hope to test the range of perceptual responses to turbine-generated infrasound and audible sound. [Work supported by Xcel Energy RD4-12 to Jeffrey Marr, St. Anthony Falls Laboratory.]

9:05

4aNS4. Effects caused by sounds at low frequencies. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de), Brigitte Schulte-Fortkamp (TU Berlin, Inst. of Fluid Mech. and Eng. Acoust., Berlin, Germany), Wade Bray (Acoust., Inc., Brighton, MI), and Klaus Genuit (Acoust. GmbH, Herzogenrath, Germany)

A central tenet of the Soundscape concept is that humans immersed in sonic environments are as "new experts" reliable communicators, whose reports and descriptions of sound and its effects must be taken seriously. As a first step, different experiments focusing on wind turbines coupled with detailed questionnaires were carried out to study the link between specific characteristics of wind turbine noises and annoyance reactions. It seems as if this will enable to determine the benefit of the Soundscape approach for investigated low frequency noise phenomena. The challenge of validating and resolving widespread and growing reports of health, physiological and behavioral effects apparently from new environmental infrasound can be met through coupling soundscape with parallel scientific techniques not limited to acoustic measurements. The soundscape concept has an incipient role to connect and approach the resolution of this complex issue. Such an augmented soundscape approach centering on the sensitivity of human beings is as important and applicable to responses to effects from sound as it is to responses to directly audible sound. This is a new kind of merged sound quality and public health issue combining acoustic, psychoacoustic, and medical aspects. Its resolution depends on a combined soundscape-mediated approach.

9:25

4aNS5. Patient Centered Medicine and Soundscape—A bridge between clinicians and acousticians. Robert Y. McMurtry (Surgery, Western Univ., 403 Main St., Picton, ON K0K2T0, Canada, rymcmurtry1@gmail.com)

The Patient Centred Method (PCM) and Soundscape have much in common including their emergence about 60 years ago based on the work of Balint and Kryter, respectively. Both place the patient or person at the center of management of clinical illness or noise annoyance. PCM requires that the patient perceive that they have experienced meaningful care, communication, and common ground in clinical encounters. The evaluation focuses on the patient's life context and their perception of disease or the "illness experience." When PCM is accomplished the result is higher satisfaction, better outcomes of chronic diseases, fewer tests and referrals, and attendant lower costs (Stewart *et al.* 2000). Soundscape, a term coined by Shafer in 1977 also places the person in center, in the context of their sonic environment, emphasizes their perception of noise as the "New Experts" (Bray 2012). According to Bray, exposed people are "objective measuring instruments whose reports and experiences must be taken seriously and quantified by technical measurements." This paper will explore the congruence of PCM and Soundscape and the applicability to environmental noise assessments. The necessity of this approach in evaluating the impact (e.g., PTSD) on those exposed to wind turbine acoustical energy will be explored.

9:45

4aNS6. Threshold of hearing versus threshold of sensation for low frequency and infrasound. Steven E. Cooper (The Acoust. Group, 22 Fred St., Lilyfield, NSW 2040, Australia, drnoise@acoustics.com.au)

Residents impacted by wind farms identify the perception of a sensation not necessarily noise. By use of a pressure sound field in a damped reverberation chamber the Threshold of Hearing for Infrasound and Low Frequency (refer Wattanbe and Moller) has been explored together with the Threshold of Sensation for sine waves. Those results are then compared using band limited pulsed signals from a Wind Farm. The differences and what they mean for Wind Turbine investigations are discussed.

10:05–10:25 Break

10:25

4aNS7. Wind farm infrasound—Are we measuring what is actually there or something else? (Part 2). Steven E. Cooper (The Acoust. Group, 22 fred St., Lilyfield, NSW 2040, Australia, drnoise@acoustics.com.au)

In predigital acoustics, low frequency analysis used analog narrow band filters and cathode ray oscilloscopes for special problems leading to the general use of peak values. Analog filters have time constants that can affect the derived rms values requiring caution where high crest factors are involved. Modern narrowband digital analysis based on a FFT of the time signal extracts the periodic function that occurs in the time domain that are then displayed as discrete peaks in the frequency domain. FFT analysis of turbines show discrete infrasound peaks at multiples of the blade pass frequency in addition to sidebands in the low frequency range spaced at multiples of the blade pass frequency. Are these signals actually there or are they a product of modern day analysis. Is the infrasound signature a clue to a different area of investigation? In Jacksonville, Part 1 presented the complexity of the investigations and showed how the raw Pascal data are lost when converting to SPL and then A-weighting. Part 2 presents the results of different filtering techniques for different wind farms.

10:45

4aNS8. An evaluation of how nightly variations in wind turbine noise levels influence wrist actigraphy measured sleep patterns. David S. Michaud (Health Canada, Canadian Federal Government, 775 Brookfield Rd., Ottawa, ON K1A 1C1, Canada, david.michaud@canada.ca)

Health Canada's Wind Turbine Noise and Health Study assessed self-reported and objective measures of sleep on a sub-sample of the study's 1238 participants. The data analysis indicated that calculated long term outdoor wind turbine noise (WTN) levels up to 46 dBA did not have a significant influence on the evaluated measures of sleep (Michaud *et al*., 2016, Sleep **39**, 97–109). A more refined analysis is being conducted to assess wrist actigraphy measured sleep patterns in relation to nightly variations in wind turbine operations. Variations in turbine operations (i.e., RPM and electrical power) were used to calculate WTN levels in 10-min intervals and time-synchronised with sleep watch data collected in 1-min epochs. The 10-min sound levels and daily sleep diaries (relied upon to adjust for closed or open windows) were used to estimate indoor A-weighted WTN levels. The correction factor used to obtain indoor sound levels was derived from a series of field measurements designed to investigate the outdoor to indoor sound pressure level difference on a representative sample of dwellings. The analysis is restricted to participants living between 0.25 and 1 km from wind turbines (116 males and 159 females). At these distances, WTN levels are the highest making it more likely to detect potential WTN impacts on sleep. Results will be presented for multiple measures of sleep in 10-min intervals and nightly averages for up to seven consecutive sleep nights.

11:05

4aNS9. Wind turbine acoustic analysis should be performed to minimize the potential for adverse impacts on life-quality for neighbors. Robert W. Rand (Rand Acoust., 1085 Tantra Park Circle, Boulder, CO 80305, rrand@randacoustics.com) and Stephen E. Ambrose (S.E. Ambrose & Assoc., Windham, ME)

Wind turbine environmental assessments should be expanded to include evaluating infrasonic, low-frequency, and A-weighted noise levels. Modern instruments and microphones can measure three simultaneous, band-pass measurements: dBL, dBC, and dBA. This paper will evaluate these band-pass measurements and discuss potentials for predicting potential adverse public responses.

11:25

4aNS10. Reproducing wind farm infrasound for subjective testingJust how accurate is the reproduced signal? Steven E. Cooper (The Acoust. Group, 22 Fred St., Lilyfield, NSW 2040, Australia, drnoise@acoustics.com.au)

Apparently on the basis of room modes being excited in residential dwellings, one concept has been to ignore high quality field measurement recordings (wave files) and use the narrow band (FFT) Leq results of a 10 min sample to create, by superimposing a number of sine waves, a trapezoidal time signal as the source for subjective testing and restricting the bandwidth to only infrasound. Other testing utilizes full spectrum signals but has limitations on having an accurate signal due to the limitation of the speakers. An examination of both methods and the limitations of the results have been examined and will be discussed.

Session 4aPA

Physical Acoustics: Multiple Scattering I

Valerie Pinfield, Cochair
Chemical Engineering Department, Loughborough University, Loughborough LE11 3TU, United Kingdom

Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

Chair's Introduction—8:00

Invited Papers

8:05

4aPA1. Integrated extinction, or total scattered energy, as a linear sum of individual contributions in acoustic multiple scattering. Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Multiple scattering (MS) involves interaction between scatterers with the result that the scattered field is not simply the sum of the separate responses. In fact, it is generally a very nonlinear process, particularly at high frequencies. It is therefore surprising that there are circumstances for which the total energy of the scattered acoustic wave is simply the sum of the individual contributions. Total scattered energy, also known as the integrated extinction, is defined as the integral of the scattering cross-section with respect to wavelength for a given direction of incidence, and is therefore a measure of the scattering over all frequencies. Two conditions are necessary, one at zero frequency and the other at infinite frequency. They are, respectively, (i) that all scatterers have effective static compressibility and density equal to the background properties, and (ii) that the wavefront in the forward direction associated with MS does not precede the incident wavefront. Examples will be given showing how both conditions can be achieved with regular and irregular arrays of elastic shells. This linear MS result is particular to acoustics, as compared with electromagnetic waves *in vacuo*. [Work supported by ONR.]

8:25

4aPA2. The effective material properties of a composite elastic half-space. Ian D. Abrahams and William J. Parnell (School of Mathematics, Univ. of Manchester, Manchester M13 9PL, United Kingdom, i.d.abrahams@manchester.ac.uk)

A classical problem in applied mathematics is the determination of the effective properties of a composite material by looking at its reflection and transmission properties. This model discussed here is an elastic half-space containing randomly distributed voids—to obtain its "average" material properties (i.e., the effective density and elastic moduli) we consider elastic waves incident from a homogeneous half-space onto the inhomogeneous material. We restrict attention to dilute dispersions of inclusions, and therefore, results are obtained under the assumption of small volume fraction. We look at several aspects of this problem, as times allows. First, we discuss how predictions derived from the non-isotropic Foldy or the Waterman-Truell multiple-scattering theories (MSTs) in the low-frequency limit are equivalent to results found by an asymptotic integral equation (homogenization) method developed by the authors [1]. Second, the effect of nonlinear pre-stress on each void, and hence on the averaged material properties, is considered [2]. And third, some comments are offered regarding higher order effects and the closure assumption (e.g., the quasi-crystalline approximation). [1] W. J. Parnell, I. D. Abrahams, and P. R. Brazier-Smith, Quart. J. Mech. Appl. Math. **63**, 145–175, 2010. [2] T. Shearer, W. J. Parnell, and I. D. Abrahams, **471**, 20150450, 2015.

8:45

4aPA3. Full transmission and reflection of waves propagating through a maze of disorder. Benoît Gérardin, Jérôme Laurent, Arnaud Derode, Claire Prada, and Alexandre Aubry (ESPCI ParisTech, PSL Res. Univ., CNRS, Université Paris Diderot, Sorbonne Paris Cité, Institut Langevin, 1 rue Jussieu, Paris 75005, France, benoit.gerardin@espci.fr)

Multiple scattering of waves in disordered media is often seen as a nightmare whether it be for communication, imaging, or focusing purposes. The ability to control wave propagation through scattering media is thus of fundamental interest in many domains of wave physics, ranging from optics or acoustics to medical imaging or telecommunications. Thirty years ago, it was shown theoretically that a properly designed combination of incident waves could be fully transmitted through (or reflected by) a disordered medium. Although this remarkable prediction has attracted a great deal of attention, open and closed channels have never been accessed experimentally. Here, we study the propagation of elastic waves through a disordered elastic waveguide. Thereby, we present experimental measurements of the full S-matrix across a disordered elastic wave guide. To that aim, laser-ultrasonic techniques have been used in order to obtain a satisfying spatial sampling of the field at the input and output of the scattering medium. The unitarity of the S-matrix is investigated and the eigenvalues of the transmission matrix are shown to follow the expected bimodal distribution. Full transmission and reflection of waves propagating through disorder are obtained for the first time experimentally. The wave-fields associated to these open and closed channels are imaged within the scattering medium to highlight the interference effects operating in each case.

9:05

4aPA4. Stochastic theory for multiple scattering of ultrasound in particulate media. Felix Alba (Felix ALBA Consultants, Inc., 1159 Sunset Dunes Way, Draper, UT 84020, felixalba@q.com)

Even though there have been published/used several approaches to modeling the complex phenomenon of *multiple scattering*, they are either of a *semi-empirical* nature with severe limitations in size/wavelength range and performance or, albeit having a *fundamental flavor*, have proven not accurate enough to be employed in broad range/application scientific instruments. Fundamental versus phenomenological approaches are discussed. The METAMODEL™ fundamental multiple-scattering theory is as fundamental as Mie (optics) and ECAH (acoustics) single-scattering theories. Besides its *fundamental* character, its *generic* validity resides in having described and calculated the detailed interaction between the scattered fields produced by every particle, treating such an interaction in a statistical sense and leading to *generic stochastic field equations* where the overlapping of all scattered fields is equally contemplated regardless of their physical nature (viscous-inertial, thermal diffusion, elastic, electromagnetic, etc.). The fundamentals of this theory are described in some mathematical detail. Using the Ultra-SCATTERER™ R & D software tool, predicted as well as experimental data in concentrated suspensions, emulsions, and aerosols (collected with commercially available spectrometers) are presented.

9:25

4aPA5. Acoustic propagation and multiple scattering in powders. Malcolm J. Povey, Mel Holmes, and Raied Al-Lashi (School of Food Sci. and Nutrition, Univ. of Leeds, Leeds, West Yorkshire ls2 9jt, United Kingdom, m.j.w.povey@leeds.ac.uk)

Determination of the material properties of powders is critical in very many industries yet crucial transformations such as the transition from a free-flowing powder to a caked powder defy quantitative measurement. In particular, distinguishing between the onset of caking (so that it may be prevented, for example) and consolidation or segregation of powders which have not caked is very difficult. We present here calculations of the acoustic properties of free flowing powders based on scattering theory (Alba, 2004), together with model calculations for caked and consolidated powders within which an effective stiffness has developed (Coghill 2011, Makse 2004). These calculated values are compared with the measured acoustic properties of silica powders demonstrating that acoustic measures such as the velocity of sound and the attenuation of sound (insertion loss) are highly effective in distinguishing between caking and consolidation. Alba. 2004, Chapter 10 of "Concentrated dispersions, theory, experiments, and applications," edited by P. Somasundaran and B. Markovic, American Chemical Society (ACS) Symposium Series 878. Coghill and Giang, Powder Technol. **208**, 694–701, 2011. doi:10.1016/j.powtec.2010.11.040. http://dx.doi.org/10.1016/j.powtec.2010.11.040. Makse, Gland, Johnson, and Schwartz, Phys. Rev. E**70**, 1–19, 2004. doi:10.1103/PhysRevE.70.061302.

9:45–10:05 Break

Contributed Papers

10:05

4aPA6. Nonlinear acoustic forces acting on inhomogeneous fluids at slow time-scales. Jonas T. Karlsen (Dept. of Phys., Tech. Univ. of Denmark, DTU Phys., Bldg. 309, Kongens Lyngby 2800, Denmark, jonkar@fysik.dtu.dk), Per Augustsson (Dept. of Biomedical Eng., Lund Univ., Lund, Sweden), and Henrik Bruus (Dept. of Phys., Tech. Univ. of Denmark, Kongens Lyngby, Denmark)

We present a novel theory describing the nonlinear acoustic force density acting on a fluid of inhomogeneous density and compressibility, for example, due to an added salt concentration. We derive an expression for the time-averaged acoustic force density acting on an inhomogeneous fluid, which depends on the gradients of the fluid density and compressibility. This smeared-out force density can be interpreted as a generalization of the well-known acoustic forces acting on a particle or an immiscible-fluid interface. The special case where the speed of sound in the solution is independent of the salt concentration, which is a good approximation for many actual salt solutions, leads to a particularly simple theoretical description. The theory predicts that in microfluidic channels, the nonlinear acoustic forces act to relocate density distributions into field-dependent configurations, which are stabilized against gravitational collapse driven by hydrostatic pressure gradients. We show the first experimental confirmations of these predictions obtained by confocal imaging in glass-silicon microchips.

10:20

4aPA7. Multiple scattering in bubbly media: Highlighting the role played by interactions between neighboring bubbles. Maxime Lanoy (Institut Langevin, 1, rue jussieu, Paris 75005, France, maxime.lanoy@espci.fr), Valentin Leroy (Matière et Systèmes Complexes, Paris, France), and Arnaud Tourin (Institut Langevin, Paris, France)

A bubble in water exhibits a low-frequency monopolar acoustic resonance, the so-called Minnaert resonance, which makes an assembly of such bubbles an ideal system to study multiple scattering of ultrasound. Following Foldy's seminal work, various approaches, such as the ones by Keller or Waterman and Truell, have been developed to infer the effective acoustic properties of a bubble cloud. Here, we confront the predictions of these different approaches with numerical results obtained with a multiple scattering theory (MST) code that fully incorporates the multiple scattering effects. Based on this study, we show the importance to take into account the interactions between neighboring bubbles to predict the behavior of a bubble sample. Finally, we demonstrate how the introduction of a local order can affect the effective parameters and allows interesting transmission properties.

10:35

4aPA8. Acoustic harmonic generation and phase conjugation with a single layer of bubbles. Olivier Lombard (Univ. Paris Diderot, 10 rue Alice Domon et Leonie Duquet, Paris 75205 Cedex 13, France, olivier.lombard@univ-paris-diderot.fr), Christophe Barriere (Institut Langevin, Paris, France), and Valentin Leroy (Matière et Systèmes Complexes, Paris, France)

Bubbles are well known for being strong nonlinear scatterers in acoustics. This feature is actually used in medical ultrasound, and nonlinear phenomena, such as phase conjugation [1] or second-harmonic generation [2], had been reported in bubbly liquids. We carried out experiments and calculations on a particularly simple bubbly medium: a single layer of bubbles trapped in a yield-stress fluid. We show the existence of an ideal bubble concentration that maximizes the nonlinear second harmonic generation by the layer. It results from the interplay between the nonlinear response and the strong multiple scattering in the system [3]. The same nonlinear mechanism can be used to obtain phase conjugation, with a probe wave at frequency f, and pump wave at 2f. The single layer of bubbles is then a subwavelength phase conjugation mirror. We studied its efficiency in terms of direction and magnitude of the reflected wave. [1] D. V. Vlasov *et al*., Sov. Phys. Acoust. **29** (1983). [2] J. Wu, Z. Zhu, J. Acoust. Soc. Am **89** (6) (1991) [3] O. Lombard *et al*., EPL, **112** (2015).

10:50

4aPA9. Manipulating air bubbles with secondary Bjerknes forces. Maxime Lanoy (Institut Langevin, 1, rue jussieu, Paris 75005, France, maxime.lanoy@espci.fr), Caroline Derec (Matière et Systèmes Complexes, Paris, France), Arnaud Tourin (Institut Langevin, Paris, France), and Valentin Leroy (Matière et Systèmes Complexes, Paris, France)

Gas bubbles in a sound field are submitted to a radiative force, known as the secondary Bjerknes force. In this presentation, we propose an original experimental setup that allowed us to investigate in details this force between two neighboring bubbles. The dependance with the sonication frequency, as well as the bubbles radii and distance, was examined. We report the observation of both attractive and, more interestingly, repulsive Bjerknes force, when the two bubbles have different radii and can thus be driven in antiphase. On the contrary, our results also show the importance of taking multiple scattering into account when the bubbles radii become similar. The setup demonstrates the accuracy of secondary Bjerknes forces for attracting or repealing a bubble, and could lead to new acoustic tools for non-contact manipulation in microfluidic devices.

11:05

4aPA10. Nonstructural acousto-injection luminescence in metalized lithium niobate. Igor Ostrovskii (Phys. and Astronomy, Univ. of MS, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu), Oleg Korotchenkov, Nikolaj Borovoy, Andriy Nadtochiy, Roman Chupryna (Taras Shevchenko Univ., Kyiv, Ukraine), and Chandrima Chatterjee (Phys. and Astronomy, Univ. of MS, Oxford, MS)

The observation of a nonstructural acousto-injection luminescence (NAIL) from metallized $LiNbO_3$ wafers is reported. The X- and Y-cut plates with linear dimensions of a few mm and silver paste electrodes on opposite surfaces are investigated. The experiments are done at room temperature. The fundamental shear modes are excited at MHz-frequencies. We measure the spectra of NAIL, acousto-electric resonance/antiresonance properties, X-ray diffraction rocking curves, acoustic emission accompanying NAIL, and photoluminescence. The NAIL and associated effects appear above a certain threshold acoustical strain of $e = 10^{-5}$. The results are explained in the terms of considerable piezoelectric fields, yielding the charge injection from the metal contacts into crystal along with the strong mechanical stresses leading to dislocations motion. The acoustic emission and X-Ray rocking curves disclose the dislocation motion under $e > 10^{-5}$. The involvement of the microstructural non-uniformities in the effects observed is experimentally identified by the X-ray rocking curves taken at different ultrasound amplitudes and photo-luminescence spectra taken from the different micro-regions of samples. Photoluminescence reveals the charged point defects that may promote an electrical conduction. The distribution of crystal defects along wafers is not uniform, and has a quasi-periodical component with tens to hundreds of microns spacing between their extremal locations.

THURSDAY MORNING, 26 MAY 2016 — SALON F, 8:00 A.M. TO 11:30 A.M.

Session 4aPP

Psychological and Physiological Acoustics: Temporal Aspects of Auditory Processing (Poster Session)

Magdalena Wojtczak, Chair
Psychology, University of Minnesota, 1237 Imperial Ln., New Brighton, MN 55112

All posters will be on display from 8:00 a.m. to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9:45 a.m., and authors of even-numbered papers will be at their posters from 9:45 a.m. to 11:30 a.m.

Contributed Papers

4aPP1. Effects of age and hearing loss on coding frequency and amplitude modulation. Kelly L. Whiteford (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, whit1945@umn.edu), Heather A. Kreft (Otolaryngol., Univ. of Minnesota, Minneapolis, MN), and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

Is phase-locking to temporal fine structure (TFS) selectively affected by hearing loss or ageing? This question has clinical relevance because audiometric measures may be insensitive to TFS deficits linked to auditory neuropathy/dyssynchrony and synaptopathy (“hidden hearing loss”). Previous work has suggested that slow-rate FM is coded via phase-locking to TFS, whereas fast-rate FM is converted to amplitude modulation (AM) via cochlear filtering. This hypothesis was tested by correlating performance in slow- and fast-rate FM with performance in tasks known to reflect TFS coding (interaural-time-difference detection, ITD) and cochlear filtering (forward masking patterns). Subjects with clinically normal hearing at low frequencies between the ages of 20 and 80 years were tested, with approximately 10 subjects per decade. Effects of low- and high-frequency hearing loss were also examined, while controlling for age. Although correlations were found between all measures of FM and AM, no clear specific effect of age was found on TFS coding (independent of AM coding). Preliminary results with hearing-impaired listeners suggest an effect of cochlear filtering that may not be specific to the fast-rate FM detection. Overall, the results do not provide strong evidence for TFS-specific deficits with either age or hearing loss. [Work supported by NIH grant R01DC005216.]

4aPP2. Spike-timing and mean-rate coding of the temporal fine structure and envelope cues in real words. Ian C. Bruce and Michael R. Wirtzfeld (Elec. and Comput. Eng., McMaster Univ., Rm. ITB-A213, 1280 Main St. W, Hamilton, ON L8S 4K1, Canada, ibruce@ieee.org)

A number of studies over the past decade have argued for the importance of temporal fine structure (TFS) cues for the perception of consonants. However, recent investigations indicate that TFS cues from consonants may largely be converted into envelope (ENV) cues by narrowband cochlear filtering, such that these cues are conveyed by the mean-rate response of auditory nerve fibers rather than spike-timing cues. However, these studies used nonsense VCV syllables, and this result may not generalize to real words in which the patterns of ENV and TFS cues may be substantially different, and in which lexical context may play a role. In this study, we used a computational model of the auditory periphery and neural-based speech intelligibility predictors to investigate the TFS and ENV representation of real words from the NU-6 database. Spike-timing and mean-rate cues were evaluated for "auditory chimaeras" created from this database, in which the TFS of one signal is mixed with the ENV of another. The results indicate that the chimaera processing has a bigger impact in general on the mean-rate representation of phonemes than on the spike-timing representation, and inclusion of the spike-timing cues gives better predictions of phoneme perception. [Funded by NSERC of Canada.]

4aPP3. Cues in detection of rhythmic irregularities. Magdalena Wojtczak (Psych., Univ. of Minnesota, 1237 Imperial Ln., New Brighton, MN 55112, wojtc001@umn.edu) and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

A recent study [Hove *et al.* (2014). *Proc. Natl. Acad. Sci.* **111**, 10383–10388] showed that the perception of rhythm carried by simultaneously presented low-pitch and high-pitch complex tones is more easily disrupted by timing irregularities in the low tone than in the high tone. This result led the authors to conclude that low-pitch tones are used for laying down the rhythm in music because they are more precise rhythmic time markers than high-pitch tones. In this study, no difference was found in the temporal acuity for low and high tones. Instead, asynchrony detection was found to be better for low-leading than for high-leading tone pairs, regardless of which tone was rhythmically irregular in a sequence. The results are consistent with an asymmetry in the perception of asynchrony between a low and high tone pair without the rhythmic context [Wojtczak *et al.* (2012). J. Acoust. Soc. Am. **131**, 363–377]. The outcome leads to a reinterpretation of the results of Hove *et al.*, based on asynchrony, rather than rhythm, perception. Possible ecological reasons for the asymmetry in asynchrony perception include the natural properties of both sound sources and the peripheral auditory system. [Work supported by NIH grant R01 DC005216.]

4aPP4. Psychophysiological responses to listening to speech in intermittent noise. Alexander L. Francis (Purdue Univ., SLHS, Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, francisa@purdue.edu)

Listening to speech in noise can be effortful, and the presence of background noise may in itself provoke physiologically measurable stress. Typical laboratory tests of speech perception in noise often present masked stimuli in trials separated by silence, in effect starting and stopping an otherwise constant background noise multiple times over the course of an experiment. While these interruptions might reduce listener stress by providing momentary respite from an aversive stimulus, intermittent noise might also increase stress by repeatedly inducing automatic physiological responses to the noise onsets. Here, I present the results of an experiment contrasting these possibilities. Younger (age 18–36) and older (age 60 +) listeners heard sentences presented in speech-shaped noise at 0 dB SNR while physiological responses linked to stress and arousal (skin conductance, heart rate, fingertip pulse amplitude, and facial electromyography) were recorded. Two roughly 15-min blocks of noise, each containing 36 unique sentences, were presented. In the interrupted noise condition the noise was silenced for 5 s shortly after each sentence while in the uninterrupted condition the noise continued unabated. Behavioral measures of listening task performance and physiological measures collected during listening and speaking will be presented, and implications for future research will be discussed.

4aPP5. The role of a temporal mechanism in the perception of speech-like logarithmic frequency sweeps. Carolyn M. McClaskey (Cognit. Sci., Univ. of California, Irvine, 4308 Palo Verde Rd., Irvine, CA 92617-4321, carolyn.mcclaskey@gmail.com), Daniel Cramer (Oberlin College, Oberlin, OH), and Kourosh Saberi (Cognit. Sci., Univ. of California, Irvine, Irvine, CA)

The auditory system is thought to process dynamic changes in frequency via two complementary mechanisms: a phase-locking-based mechanism that is limited to slowly changing stimuli at low (<4–5 kHz) frequencies, and an energy-based mechanism that functions across all frequency regions and all rates of change. The current study investigates the relative contribution of these two mechanisms—and the role of phase-locking cues in particular—in the perception of low, slow frequency sweeps that closely parallel those found in speech prosody. Listeners identified the direction of unidirectional logarithmic frequency sweeps in a single-interval identification task. Both the rate and extent of frequency change (i.e. transition span) were uniformly varied over a range of 0.0147–0.1667 octaves/s and 0.1–0.5 semitones, respectively. Stimuli were roved around a center frequency of 500 Hz, and all were between 50 and 1000 ms in length. Results show that sensitivity (d') significantly increases with increasing transition span but significantly decreases with increasing rate of frequency change, suggesting that phase-locking cues may contribute to the perception of sweeps with very slow rates of change. Data were well-predicted by a multivariate linear model as a joint function of sweep rate and transition span.

4aPP6. Limitations on temporal processing by cochlear implant users. Robert P. Carlyon, Stefano Cosentino, John M. Deeks (Medical Res. Council, Cognition & Brain Sci. Unit, MRC CBU, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, bob.carlyon@mrc-cbu.cam.ac.uk), Wendy Parkinson (Speech & Hearing Sci., Univ. of Washington, Seattle, WA), and Julie A. Bierer (Speech & Hearing Sci., Univ. of Washington, Washington, WA)

Two experiments studied the deterioration in rate discrimination and pitch perception for pulse trains presented to single electrodes at high pulse rates. The first measured rate discrimination DLs ("RDLs") for 100-pps and 400-pps standard rates, for each of 4–5 electrodes and for 10 Advanced Bionics cochlear implant users. Thresholds were measured using two interleaved adaptive tracks, corresponding to the 100- and 400-pps standard rates. Gap detection thresholds ("GDTs") for a 1031-pps pulse train were also measured. There was a highly significant across-subject correlation between GDT and the 400-pps but not the 100-pps RDL, and these two correlations differed significantly from each other. Similarly, the across-electrode correlation between GDT and the 400-pps RDL was marginally significant, whereas there was no correlation between GDT and the 100-pps RDL. These findings are consistent with the deterioration in high-rate temporal processing sharing a common basis with the mechanisms involved in gap detection, but not with the limitations in low-rate temporal processing. We will also report the results of a second experiment that measured rate discrimination and pitch ranking at low and high rates, both on the same day that patients' implants were activated and after two months of listening experience.

4aPP7. Deficits in the sensitivity to pitch sweeps by school-aged children wearing cochlear implants. Mickael L. Deroche (Ctr. for Res. on Brain, Lang., and Music, McGill Univ., Rabinovitch House, 3640 rue de la Montagne, Montreal, QC H3G 2A8, Canada, mickael.deroche@mcgill.ca), Aditya M. Kulkarni, Julie A. Christensen (Auditory Prostheses and Percept. Lab., Boys Town National Res. Hospital, Omaha, NE), Charles J. Limb (Dept. of Otolaryngol. – Head and Neck Surgery, Univ. of California San Francisco School of Medicine, San Francisco, CA), and Monita Chatterjee (Auditory Prostheses and Percept. Lab., Boys Town National Res. Hospital, Omaha, NE)

Sensitivity to static changes in pitch has been shown to be poorer in school-aged children wearing cochlear implants (CIs) than children with normal hearing (NH), but it is unclear whether this is also the case for dynamic changes. Yet, dynamically changing pitch has considerable ecological relevance in terms of natural speech, particularly aspects such as intonation, emotion, or lexical tone information. This study examined children and

adults, with NH or wearing a CI using clinically assigned settings with envelope-based coding strategies. Percent correct was measured in one- or three-interval two-alternative forced choice tasks, for the direction or discrimination of harmonic complexes based on a linearly rising or falling fundamental frequency. Sweep rates were adjusted per subject, in a logarithmic scale, so as to cover the full extent of the psychometric function. Data for up- and down-sweeps were fitted separately, using a maximum-likelihood technique. Hits and false alarms were then converted into d' and beta values, from which a threshold was extracted at a d' of 0.77. Thresholds were very consistent between the two tasks and considerably higher (worse) for CI listeners than for their NH peers. Thresholds were also higher for children than adults. Factors such as age at implantation, age at profound hearing loss, and duration of CI experience did not play any major role in this sensitivity. Sweep direction thresholds held the most predictive power for performance in tasks related to speech prosody.

4aPP8. Neural activation patterns indicate robust auditory nerve synchrony generated with octave-band chirps. Ivy Thompson and Brian Earl (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave, Cincinnati, OH 45267, thompsiv@mail.uc.edu)

Recent research has suggested that the amplitude of high-intensity compound action potentials (CAPs) can detect auditory nerve damage that is missed by traditional threshold measurements. The use of octave-band chirps may enhance neural synchrony and provide clinicians and researchers a precise tool for identifying changes in auditory nerve integrity. A group of Mongolian gerbils (N = 12) was used to compare the difference in amplitude-intensity functions and neural activation patterns generated by octave-band chirps and tonebursts at octave frequencies between 2 and 16 kHz at 40, 60, and 80 dB SPL. Amplitude-intensity functions revealed larger absolute CAP amplitudes for octave-band chirps than for tonebursts. Neural activation patterns were constructed by plotting the derivative of CAP amplitude across multiple conditions of simultaneous broadband noise that was high passed in 1/3 octave intervals. Neural activation patterns for octave-band chirps and tonebursts showed that the frequency location of peak activation was generally equivalent while the height of peak activation was greater for octave-band chirps than for tonebursts. This suggests that octave-band chirps elicit more synchronous neural firing, thereby indicating that they could be an optimal stimulus for detecting regional damage in auditory neurons that encode supra-threshold stimuli.

4aPP9. Effects of elevated amplitude modulation discrimination threshold on simultaneous amplitude modulation rate discrimination. Sean R. Anderson, Alan Kan, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, Office 564, 1500 Highland Ave., Madison, WI 53705, sean.anderson@wisc.edu)

Cochlear-implant (CI) users struggle to understand speech in difficult listening environments partly because they have limited access to auditory cues that allow for perceptual segregation between target and competing sounds. Because CI speech coding strategies optimize the encoding of envelope cues to promote better speech intelligibility, envelope may be an important cue for source segregation in CI users. Recent work has suggested that access to amplitude modulation (AM) depth and rate varies according to location of electrodes in the cochlea, and that turning off electrodes that yield poorer AM sensitivity may improve speech reception. The purpose of this study was to investigate the role of AM sensitivity in the ability of listeners to use AM rate information in multiple electrodes. It was hypothesized that electrodes with higher AM thresholds, which were simulated in normal-hearing listeners, would limit listeners' ability to segregate AM rates. Subjects discriminated between pairs of stimuli that varied in AM rate relative to a reference rate. Stimulus pairing was within- or across-ears. Results suggest that AM insensitive electrodes impair CI users' ability to discriminate AM rates, which may limit source segregation using envelope cues in complex listening environments. [This work was supported by NIH-NIDCD R01 DC003083 (Litovsky).]

4aPP10. Effects of a precursor on amplitude modulation detection are consistent with efferent feedback. Ali Almishaal and Skyler G. Jennings (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1217 BEH S, Salt Lake City, UT 84112, ali.almishaal@utah.edu)

The acoustic waveform of speech is characterized by slowly varying amplitude fluctuations (i.e., envelope) and an accurate representation of the envelope is essential for speech understanding. The post-cochlear representation of the contrast between peaks and valleys of the envelope (peak-to-valley contrast) may be reduced by cochlear compression. This study tested (1) whether amplitude modulation (AM) detection thresholds are consistent with cochlear compression and (2) whether the introduction of a precursor before the carrier results in improved AM thresholds, consistent with a decompressed cochlear response via the medial olivocochlear reflex (MOCR). In the no precursor condition, AM thresholds worsen at mid-levels, consistent with reduced peak-to-valley contrast from cochlear compression. In the precursor condition, AM thresholds improved at low modulation frequencies and mid-to-high levels, consistent with a reduction in cochlear amplifier gain and decompression of the cochlear input/output function via the MOCR. These findings suggest the MOCR may play a role in the perception of other amplitude-modulated stimuli, such as speech.

4aPP11. How well do measures of the precedence effect based on clicks predict performance for long-duration stimuli? M. Torben Pastore and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., 4 Irving Pl., Troy, NY 12180, m.torben.pastore@gmail.com)

Despite multiple reflections off nearby surfaces, listeners often localize sound sources based primarily upon the first arriving, direct sound. This is called the precedence effect. Much has been learned about the precedence effect using transient clicks, but the vast majority of everyday sounds are relatively long in duration. Recently, Pastore and Braasch (JASA, 2015) tested the effects of increased lag intensity on localization dominance for longer, 200-ms duration stimuli with 20-ms cosine-squared ramps. Assuming that interactions at the onset of lead-lag stimuli are primarily responsible for the precedence effect, it has been suggested that the binaural cues important to the precedence mechanism are the same for clicks and longer-duration noise stimuli. To test this hypothesis, we presented lead-lag stimuli composed of 1-ms, rectangular clicks, as well as 41-ms and the previously used 200-ms long noise bursts. Five, lead-lag delays between 1 and 5 ms were tested for lead/lag level differences of 0-, –4-, and –8-dB. Results and statistics clearly show that the classic click stimuli and long-duration stimuli do not yield the same performance for the conditions tested in these experiments, especially when lag level is increased. [Research supported by NSF 1320059.]

4aPP12. Backward masking determination with simultaneous early, middle, and late evoked potentials. Silas Smith, Robert Sears, and Al Yonovitz (Dept. of CSD, The Univ. of Montana,Missoula, MT 59812, al.yonovitz@umontana.edu)

Backward masking (BM) functions have been shown to relate to age, lead toxicity, and are differentiated in children with language disorders. These functions may be indicative of auditory processing deficits. This study investigated if evoked potentials (EP) could be utilized to obtain BM functions. A tonal stimulus, followed by an ISI and a noise masker was the EP stimulus. All were studied individually in the appropriate temporal alignment. The design of this study allowed observation of the early, middle, and late auditory evoked potentials. This study randomly presented four different stimulus conditions: (1) tone alone, (2) noise alone, (3) tone and noise, and (4) silence. With a long inter-trial interval (1 s) and high sample rate (25600 Hz) EP's were obtained for 4000 trials. The stimuli were pure-tones (1000 Hz, 10 ms duration with a Blackman function and noise bursts of varying intensity and varied ISI. Results indicated that EP's could be arithmetically combined to observe the differential electrophysiological responses and neurologic loci of evoked potentials during the BM effect.

4aPP13. The inter-stimulus interval critical value in backward masking testing. Robert Sears, Silas Smith, Sarah Schied, and Al Yonovitz (Dept. of CSD, The Univ. of Montana, Missoula, MT 59812, robert.sears@umontana.edu)

Backward masking (BM) has been studied both as a psychoacoustic phenomenon and as a potential diagnostic indicator of auditory processing difficulty. In BM assessment, the subject responds to a brief tonal signal followed by an inter-stimulus interval (ISI) of silence and then by a noise masker. In all previous studies, an adaptive auditory threshold is obtained at each ISI. This study used unique instrumentation that allowed the adaptive ISI to maintain threshold. Twelve subjects were utilized. All had normal hearing with an age range of 18–30 years. The critical ISI was varied with an adaptive procedure in 2 ms steps. The critical ISI value was obtained for stimulus intensity levels above the tonal threshold found in a longer ISI condition. This procedure allowed less fatigue and provided an easier task for the subject compared to the tonal intensity tracking task.

4aPP14. Discrimination thresholds for level increments in the stop-consonant noise bursts of CVC words: Intra-speech masking, temporal-position, temporal-variability, and place-of-articulation effects. Blas Espinoza-Varas (Commun. Sci. & Disord., OU Health Sci. Ctr., 1200 N. Stonewall Ave., Oklahoma City, OK 73117-1215, blas-espinoza-varas@ouhsc.edu) and Jaremiah Hilton (Biostatistics and Epidemiology, OU Health Sci. Ctr., Oklahoma City, OK)

Estimates of sensitivity to level differences in stop-consonant noise bursts could help designing speech-processing strategies such as dynamic-range compression and optimal consonant-vowel intensity ratio. In normal-hearing participants, we measured level-discrimination thresholds (LDTs) for the noise bursts of CVC words (/pæt/, /pæk/, and /kæt/). With /pæt/, a 2I-2AFC task measured LDTs for the pre- or post-vocalic burst in isolation or in word context; in the latter, the burst with level increments (pre or post) remained the same or varied unpredictably from trial to trial. In isolation, the 1.98–2.22 dB LDTs approached those of like-duration random noise, but increased to ≈9.0 dB for the pre-vocalic burst in-context, and more so in unpredictable-burst conditions (10.81 dB); for the post-vocalic burst the LDTs increased only slightly. To assess the role of set-size, within-trial standard, and place-of-articulation, the in-context, unpredictable-burst LDTs were measured with single- or two-observation interval 2AFC tasks presenting only /pæt/ or, randomly, /pæt/, /pæk/, or /kæt/. The context, temporal-position, predictability, and place-of-articulation effects were significant but those of within-trial standard and set size were not; for pre-vocalic bursts, LDTs were much higher (9.97 dB) for the velar than the labial consonant (4.78 dB).

4aPP15. Comparison of Blackman, linear rise-fall, and linear rise-fall chirp signals in backward masking. Robert Sears, Silas Smith, and Al Yonovitz (Dept. of CSD, The Univ. of Montana, Missoula, MT 59812, robert.sears@umontana.edu)

Backward masking is finding applications in clinical testing and the diagnosis of auditory processing disorders. Masking of a tonal signal by a band-limited noise in a backward masking paradigm requires significant vigilance and produces fatigue. This study compared the backward masking functions of ten normal hearing young (18–30 years) subjects using tonal stimuli with different signatures in order to provide a better contrast to the noise masker. Subjects adaptively tracked the tonal threshold using 2 dB changes. The noise stimulus was 70 dB HL, and the inter-stimulus interval (ISI) varied from 2 ms to 64 ms. The results will be described in terms of fatigue, tracking efficiency, and latency.

4aPP16. Backward masking of vowel-consonant stimuli. Allie Cope, Kendra Foster, Robert Sears, Silas Smith, and Al Yonovitz (Dept. of CSD, The Univ. of Montana, Missoula, MT 59812, allie.cope@umontana.edu)

Backward masking (BM) has shown differential effects with age and auditory processing. BM may be related to reduced speech discrimination. Ten normal hearing subjects (18–30 years) were subjects in this study. This study utilized 21 VC stimuli followed by a white noise masker. The ISI interval was 5 ms, and the noise duration was 50, 100, and 200 ms. Each stimulus was randomly presented ten times. Confusion matrices determined consonant intelligibility and information transmission for distinctive features. Thus, the study investigated the accuracy of perception for English consonants as well as determined which of the distinctive features (voicing, nasality, continuancy, sibilancy, frontness, sonorancy, and labiality) remained preserved or changed when exposed to backward masking listening conditions. The results indicated a reduction in these selected features as well as reduced consonant identification as BM became more effective.

4aPP17. Backward masking and tonal audibility before and after a noise burst. Robert Sears, Silas Smith, and Al Yonovitz (Dept. of CSD, The Univ. of Montana, Missoula, MT 59812, robert.sears@umontana.edu)

This study reports on the perceived position of the tonal signal during a threshold tracking procedure. The stimulus was a 1000 Hz sinusoid tone. The inter-stimulus (ISI) interval varied from 2 ms to 64 ms. The noise stimulus was a 70 dB HL band-limited white noise. The subject tracked threshold (2 dB steps). In these experiments, ten normal hearing subjects (18–30 years) were presented the stimulus before the noise burst. Moving through a critical ISI value, the stimulus becomes inaudible after a further reduction in ISI, the stimulus is detected but appears to follow the noise. This finding supports a hypothesis that the tonal signal may be modulated by physiological loci by changing the ISI and the tonal intensity.

4aPP18. Sound recognition depends on real-world sound level. Sam V. Norman-Haignere and Josh H. McDermott (Brain & Cognit. Sci., MIT, 43 Vassar St., Rm. 4141, Cambridge, MA 02139, svnh@mit.edu)

How does the auditory system recognize instances of the same sound class with distinct acoustic properties? As a case study, we investigated the recognition of environmental sounds at different levels. In principle, level-invariant recognition could be achieved by a normalization mechanism that removes variation in level from listeners' representation of sound identity. Alternatively, listeners could use level as a cue to aid their recognition, taking advantage of the fact that different sound classes are typically heard at different levels. The latter hypothesis predicts that sounds heard at atypical levels should be more difficult to recognize. We tested this prediction by asking human listeners to identify 300 environmental sounds, each presented at seven different sound levels between 30 and 90 dB SPL. We grouped these 300 sounds into those typically heard at low (e.g. salt-shaker) and high sound levels (e.g., jackhammer) using ratings collected via Mechanical Turk. For typically loud sounds, recognition accuracy improved monotonically with increasing experiment level. But for typically quiet sounds, recognition accuracy declined at high experiment levels (above 60 dB). These results are consistent with the hypothesis that listeners recognize individual sounds by internalizing the unique distribution of acoustic features that characterize them.

4aPP19. Evaluating whether increment detection at mid-to-high pedestal frequencies is consistent with cochlear compression. Jessica Chen and Skyler G. Jennings (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1217 BEH S, Salt Lake City, UT 84112, jessica.chen@utah.edu)

The ability of the auditory system to encode the amplitude fluctuations of a signal is important for processing complex stimuli, such as speech. A study by Florentine [Florentine *et al*., J. Acoust. Soc. Am. **81**, 1528–1541 (1987)] measured intensity discrimination for multiple frequencies as a function of stimulus level. They observed that, for high frequency stimuli, difference limens were poorer at mid levels than at higher or lower levels. This mid-level "hump" is consistent with cochlear compression and suggests that compression may limit the intensity resolution of the auditory system under certain circumstances. To test the generalizability of this interpretation, the current study measured increment detection for conditions associated with the mid-level hump, based on the assumption that increment detection and intensity discrimination are determined by similar physiological processes. This study is part of a larger series of experiments in our laboratory that test whether intensity perception is consistent with peripheral processes such as cochlear compression and efferent-mediated regulation of cochlear amplifier gain.

4aPP20. Effects of interleaved noise on speech recognition in children. Carla L. Youngdahl (Communicative Sci. and Disord., Saint Mary's College, 45 Madeleva Hall, Notre Dame, IN 46556, cyoungdahl@saintmarys.edu), Sarah E. Yoho, Rachael F. Holt (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH), Frederic Apoux (Eye and Ear Inst., The Ohio State Univ. Wexner Medical Ctr., Columbus, OH), and Eric w. Healy (Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Normal-hearing adults can isolate frequency regions containing clean speech from surrounding regions containing noise. However, children have been shown to integrate information over a large number of auditory filters, and so they may not be able to isolate frequency regions as well. To assess children's level of auditory filter independence, words were filtered into 30 contiguous 1-ERB-width bands. Speech was presented in every other band, for a total of 15 speech bands. Speech-shaped noise was then added to: all 30 contiguous bands, the 15 bands not containing speech (OFF), or the 15 bands containing speech (ON). Three age groups were tested: 10 adults, 9 older children (6–7 yr old), and 9 younger children (5 yr old). Consistent with previous findings involving consonant recognition, adults displayed large performance differences between off- and on-frequency noise (OFF vs. ON). The 6- to 7-yr-old group performed similarly to adults. In contrast, the 5-yr-old group displayed equivalent performance in the OFF and ON conditions. This indicates that, for these younger children, even noise that was mostly non-overlapping in frequency interfered with speech recognition as much as noise that was on frequency. [Work supported by NIH.]

4aPP21. Development of subcortical pitch representation in three-month-old Chinese infants. Fuh-Cherng Jeng (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W224, Athens, OH 45701, jeng@ohio.edu), Chia-Der Lin (China Medical Univ. Hospital, Taichung, Taiwan), Grant R. Hollister, John T. Sabol, Garrett N. Mayhugh (Commun. Sci. and Disord., Ohio Univ., Athens, OH), Tang-Chuan Wang, and Ching-Yuan Wang (China Medical Univ. Hospital, Taichung, Taiwan)

This study investigated the development of subcortical pitch processing, as reflected by the scalp-recorded frequency-following response, during early infancy. Thirteen Chinese infants who were born and raised in Mandarin-speaking households were recruited to partake in this study. Through a prospective-longitudinal study design, infants were tested twice: at 1–3 days after birth and at three months of age. A set of four contrastive Mandarin pitch contours were used to elicit frequency-following responses. Frequency Error and Pitch Strength were derived to represent the accuracy and magnitude of the elicited responses. Paired-samples t tests were conducted and demonstrated a significant decrease in frequency error and a significant increase in pitch strength at three months of age compared to 1–3 days after birth. Results indicated the developmental trajectory of subcortical pitch processing during the first three months of life.

4aPP22. Identification of attended speech stream using single-trial electroencephalography recording. Ala Somarowthu (Dept. of BioEng., Northeastern Univ., Boston, MA), Nai Ding (College of Biomedical Eng. & Instrument Sci., Zhejing Univ., Hangzhou, China), and Ying-Yee Kong (Commun. Sci. & Disord., Northeastern Univ., Dept. of Commun. Sci. & Disord., Northeastern University, Boston, MA 02115, yykong@neu.edu)

Selective attention differentially modulates neural responses to simultaneously presented speech streams. In this study, single-trial EEG classification was performed to identify the attended speech from a two-talker speech mixture. During EEG recordings, normal-hearing listeners paid attention to one speech stream while listening to speech mixtures. The target-to-masker ratios (TMRs) varied from −9 to +9 dB. Individual speech streams were processed with head-related transfer functions to simulate different spatial locations. Two simulated spatial conditions (0 vs. +/−90 and +45 vs. −45 degree azimuth) were tested for each TMR. Features related to (1) cross-correlation values between EEG signals and temporal envelope of each speech stream, or (2) correlation values between reconstructed speech from the EEG signals with the acoustic stimuli, were fed to the classifiers. The dimensionality of the feature vector was reduced using Principal Component Analysis. Linear Discriminant Analysis and Support Vector Machine were used to classify the EEG signals. Classifiers were trained and tested with a five-fold cross validation method on data pooled across TMRs and source locations for trial lengths from 50 s to 10 s. Average classification accuracy was 85% with a 50 s trial length and maintained high at 70% with a reduced trial length of 10 s.

Session 4aSAa

Structural Acoustics and Vibration, Architectural Acoustics, Noise, and ASA Committee on Standards: Building Isolation from Seismic and Groundborne Vibration

James E. Phillips, Cochair
Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Hasson M. Tavossi, Cochair
Physics, Astronomy, & Geosciences, Valdosta State University, 2402 Spring Valley Cir, Valdosta, GA 31602

Chair's Introduction—8:30

Invited Papers

8:35

4aSAa1. Vibration isolation of buildings for control of ground borne noise. George P. Wilson (Wilson, Ihrig & Assoc., Inc., 14 Richelle Court, Lafayette, CA 94549-4821, gpwilson2@gmail.com)

Complete vibration isolation has made it possible to locate new buildings on sites with normally unacceptable levels of ground borne noise. The principles are relatively simple but the isolation details are necessarily limited and, to be successful, the isolation bearing pads or springs must have very specific dynamic and structural properties. Since the late 1980s, there have been a number of successful building isolation projects where the intruding ground borne noise from nearby rail facilities and other sources has been reduced to the threshold of hearing or less. The design principles required include massive, high mechanical impedance foundations, very low acoustical impedance isolation bearings, and relatively high stiffness and mass structure for at least two floors above the isolation level. Typical design details and information are presented regarding successful isolation configurations for entire buildings and for box-in-box isolated spaces within the building. Also included are a discussion on isolation bearing materials which have not been successful, as demonstrated by on-site measurements, and a discussion on the mechanical, acoustical, and durability requirements for successful building isolation bearings. An additional item included is a discussion of the differences between building vibration isolation design principles for noise control and building base isolation design for seismic protection.

9:05

4aSAa2. Vibration isolation for concert hall next to busy street. James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

A new, world-class performing arts center is currently being developed in a metropolitan center in the United States. The center will include two grand performance theaters. One of these theaters will be located close to a busy surface street. Vibration measurements conducted at the undeveloped site indicated that groundborne noise from street traffic would be audible within the completed theater unless measures were incorporated into the design to reduce vibration transmitted from the street to the interior of the theater. This will be achieved by structurally separating the performance area of the theater from the surrounding structure by supporting the theater on custom designed, resilient bearing pads. This paper discusses the vibration measurements taken, the projections of groundborne noise and the vibration mitigation measures that were incorporated into the structural design of the theater building to reduce groundborne noise to meet the project design criteria for background noise.

9:35

4aSAa3. Particulate damping media and isolation of ground-borne vibrations. Hasson M. Tavossi (Phys., Astronomy, and GeoSci., Valdosta State Univ., 2402, Spring Valley Cir, Valdosta, GA 31602, htavossi@valdosta.edu)

Ground borne vibration can be dissipated by attenuation, absorption, frictional, and viscous damping before being transmitted to the buildings, by means of a suitable particulate media below the structure. We show that the band-pass behavior of particulate media for mechanical vibrations can also result in significant attenuation of vibrations for frequencies on either side of the pass-band. Sample particulate structures are subjected experimentally to the longitudinal and shear waves of different amplitude and frequencies, generated by electro-mechanical shakers vibrating in different directions. The amount of vibration damping by attenuation, resonance absorption, frictional and viscous dissipations are determined experimentally. The pressure wave and shear-waves, generated experimentally by the electro-mechanical shakers, are combined to simulate realistic ground borne vibrations. Damping media consisting of composite materials with different elastic properties are tested as tuned absorbers. The goal is to determine the damping characteristics for most efficient absorption of vibration, of different intensity and frequency. The results can be used to design tuned dampers for specific ground borne vibrations and determine material characteristics for dissipation of ground borne vibrations at large scale.

10:05

4aSAa4. Measurements of the reduction in ground vibration levels from wave barrier trench. John J. LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Where buildings are adjacent to railroad tracks, there are limited practical applications for controlling the transmission of train vibration. In critical listening spaces like auditoria or government buildings, isolation of the building structure is sometimes undertaken. However, this option is not attractive for most commercial or residential development due to the attendant costs and complications. For these properties, one option for reducing levels of ground vibration is to dig a trench between the tracks and the project to act as a wave barrier. Construction of a deep trench presents a wide variety of construction challenges, and the application of this method has been rare within the United States due to the lack of regulations and sensitivity that might demand such an approach. The authors recently had a project where a wave barrier (trench) was applied, which provided the opportunity to observe the construction of the trench and to measure the resulting reduction in ground vibration levels before and after the installation. Various aspects of this process are discussed.

THURSDAY MORNING, 26 MAY 2016 SALON J, 10:50 A.M. TO 11:50 A.M.

Session 4aSAb

Structural Acoustics and Vibration, Engineering Acoustics, and Physical Acoustics: Nuclear-Powered Thermoacoustics

James E. Phillips, Cochair
Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Hasson M. Tavossi, Cochair
Physics, Astronomy, & Geosciences, Valdosta State University, 2402 Spring Valley Cir, Valdosta, GA 31602

Invited Papers

10:50

4aSAb1. Design and fabrication of a fission-powered thermoacoustic in-core sensor. Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Appl. Res. Lab, P. O. Box 30, State College, PA 16804, sxg185@psu.edu), Robert W. Smith (Marine & Physical Acoust., Appl. Res. Lab., State College, PA), James A. Smith (Fundamental Fuel Properties, Idaho National Lab., Idaho Falls, ID), and Brenden J. Heidrich (Nuclear Sci. User Facility, Idaho National Lab., Idaho Falls, ID)

A standing-wave thermoacoustic engine with dimensions identical to an ordinary fuel rod was designed to be placed in the core of the Breazeale Nuclear Reactor on Penn State's campus. The heat necessary to produce thermoacoustics oscillations was provided by two 10 mm long by 5 mm diameter, 7.5% enriched, ^{235}U fuel pellets. Those pellets were contained within a stainless-steel finned heat exchanger that was fabricated by additive manufacturing (3-D printing). The (mass-controlled) resonator was suspended in the surrogate fuel rod using two six-armed leaf springs (spiders) that centered the resonator in the "slotted tube" and allowed longitudinal vibrations of the entire resonator that coupled the oscillatory momentum of the gas within the resonator to the surrounding light-water reactor coolant. A 2.0 MPa mixture of 25% argon and 75% helium provided a trade-off between dipole radiation efficiency, resonator length, and low onset temperature differential, to produce a frequency that was high enough to be above the dominant noise produced by coolant and ^{16}N diffusion pumps. These trade-offs were optimized using the Los Alamos DELTAEC software. [Work supported by Idaho National Laboratory and Westinghouse Electric Co. Fabrication and fueling was completed in collaboration with IST-Mirion.]

11:20

4aSAb2. Using the sounds of nuclear power. James A. Smith (Fundamental Fuel Properties, Idaho National Lab., Idaho National Lab., M/S 6188, Idaho Falls, ID 83415, james.smith@inl.gov), Brenden J. Heidrich (Nuclear Sci. User Facility, Idaho National Lab., Idaho Falls, ID), Vivek Agarwal (Human Factors, Controls and Statistics, Idaho National Lab., Idaho Falls, ID), Michael D. Heibel (Global Technol. Development, Westinghouse Electric Co., Pittsburgh, PA), Robert W. Smith (Marine & Physical Acoust., Appl. Res. Lab., State College, PA), and Steven L. Garrett (Grad. Prog. in Acoust., Penn State, State College, PA)

The generation of sound by heat has been documented as an "acoustical curiosity" since a Buddhist monk reported the loud tone generated by a ceremonial rice-cooker in 1568. Over the last four decades, significant progress has been made in understanding "thermoacoustic processes," enabling the design of thermoacoustic engines and refrigerators. We have developed and tested a thermoacoustic engine that exploits the energy-rich conditions in the core of a nuclear reactor. The heat engine is self-powered and can wirelessly transmit the temperature and reactor power by generation of a pure tone which can be detected outside the reactor. We report here the first use of a fission-

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powered thermoacoustic engine capable of serving as a performance and safety sensor in the core of a research reactor and present data from two hydrophones in the coolant (far from the core) and an accelerometer attached to a structure outside the reactor. These measurements confirmed that the frequency of the sound produced indicates the reactor's coolant temperature and that the amplitude (above an onset threshold) is related to the reactor's operating power level. These signals can be detected even in the presence of substantial background noise generated by the reactor's fluid pumps. [Work supported by Idaho National Laboratory and Westinghouse Electric Co.]

THURSDAY MORNING, 26 MAY 2016 — SALON E, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Non-Native Speech Perception and Production (Poster Session)

Melissa M. Baese-Berk, Cochair
Department of Linguistics, Michigan State University, 1290 University of Oregon, Eugene, OR 97403

Tuuli Morrill, Cochair
George Mason University, 4400 University Drive, 3E4, Fairfax, VA 22030

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors 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

4aSC1. Perception of phonological variants by non-native listeners. Hayk Abrahamyan (Psych., SUNY at Buffalo, Univ. at Buffalo, Park Hall 204, Buffalo, NY 14260, hayk@buffalo.edu)

Recent research has demonstrated that listeners process carefully pronounced words (canonical forms) more quickly and accurately than casually produced words (non-canonical, reduced forms), despite the fact that casually produced word forms are more frequent in everyday language use. To date, research on the perception of phonological variants that are typical of casually produced speech has focused, with a few exceptions, on monolingual listeners. The current research examined non-native English speakers' processing of canonical and non-canonical word-forms in an attempt to more fully understand how non-native speakers of English cope with phonological variants in American English. Monolingual American English speakers and non-native American English speakers completed a cross-modal identity priming task with canonical, non-canonical, and unrelated auditory primes and visual targets. Overall, the non-native speakers were significantly slower than native speakers at recognizing both canonical and non-canonical forms, although our data suggest that non-native speakers may encounter more specific difficulties than native speakers when processing phonological variants. Our results constitute an initial attempt at understanding how non-native speakers cope with phonological variation in their second languages.

4aSC2. Children's perception of native dialects and nonnative accents. Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu), Rachael F. Holt, Tiarah Wilcox, Sarah Mabie, and Leah Neczypor (Dept. of Speech and Hearing Sci., Ohio State Univ., Columbus, OH)

In quiet listening conditions, school-aged children can have difficulty understanding nonnative-accented speech whereas adults tend to be highly accurate. The addition of noise substantially depresses word recognition accuracy for both groups. Here, these findings are extended to the perception of an unfamiliar native dialect. Children between the ages of 5 and 7 years (n = 90) were presented with HINT-C sentences produced by three female talkers with different accents—American English (midland dialect), British English, and Japanese-accented English—in quiet or in 8-talker babble with a +4 dB SNR. Results showed highly significant main effects of accent, listening condition (noise, quiet), and age in the expected directions, as well as an interaction between talker accent and listening condition. In quiet, children showed very accurate word recognition for the American and British talkers (97% and 95% correct, respectively) with lower accuracy for the nonnative talker (73% correct). Compared to the quiet condition, performance declined more in the noise-added condition for the British (20% decline) and nonnative talkers (21%) than for the American talker (7%). These results suggest that although school-aged children can understand unfamiliar native dialects, their representations of these dialects may be fragile and highly susceptible to environmental degradation.

4aSC3. The hierarchies of phonetic realization of focus in second language speech. Ying Chen (School of Foreign Studies, Nanjing Univ. of Sci. and Technol., 200 Xiaolingwei St., Nanjing, Jiangsu 210094, China, ychen@njust.edu.cn)

Two experiments were conducted to examine the phonetic realization of focus in L2 Mandarin by L1 American English speakers and in L2 English by L1 Beijing Mandarin speakers. The production data in both experiments indicated an acoustical hierarchy of duration > intensity > F0 at the sentential level. This hierarchy was correlated with another hierarchy in sentence location relative to focus: pre-focus > in-focus > post-focus. In other words, L2 learners tended to produce more nativelike patterns of duration than intensity, and intensity than F0, to code focus. These patterns were more salient in pre-focus condition than in-focus condition, and in in-focus condition than post-focus condition. These findings are consistent with Wu and Chung (2011, ICPhS) and Chen, Xu and Guion-Anderson (2014, Phonetica) that bilingual learners used more duration and intensity than F0 and more in-focus expansion than post-focus compression to code focus in L2 speech. Post-focus compression of F0 was the most difficult acoustic cue in phonetically realizing L2 focus. The nativelikeness of focus realization in L2 speech increased with the increase of L2 experience. [This work was supported by the National Science Foundation of China #61573187 and Fundamental Research Funds for the Central Universities in China NJUSTWGY14001.]

4aSC4. Production of English vowels preceding voiced and voiceless consonants by Korean learners of English. Juyeon Chung (Linguist, Indiana Univ., Memorial Hall 322, Bloomington, IN 47405-7005, chungjulia29@gmail.com)

In English, consonant voicing has large effects on both the quality and duration of the previous vowel, as does the status of the vowel as tense or lax. Our aim is to examine whether there is L1 interference on L2 English vowel productions and whether temporal acquisition is easier than vowel quality acquisition. Korean L2 speakers were chosen as participants; Korean has no contrasts in tense vs. lax vowels, and no coda consonant voicing contrast in a monosyllabic structure, but does exhibit post-vocalic voicing contrasts in disyllabic structures, and in these cases, voiced consonants exhibit lengthening of the preceding vowel. The participants were asked to read the list of English nonce words consisting of the English high vowels and sets of plosives contrasting in voicing as a coda in the two different structures. None of the speakers exhibited different patterns in monosyllabic and disyllabic structures. All speakers did exhibit durational correlates to the voicing contrast, and to the tense-lax distinction. Formant frequency differences were found for both voicing and the tense-lax distinction, however, not as consistently. Specifically, back vowels were not distinguished by a number of subjects. In sum, temporal properties were more easily acquired than vowel quality properties.

4aSC5. Audiovisual integration in fricative production and perception by nonnative speakers. Margaret Harrison, Chao-Yang Lee (Commun. Sci. and Disord., Ohio Univ., Grover Ctr. W252, Athens, OH 45701, mh806711@ohio.edu), and Seth Wiener (Modern Lang., Carnegie Mellon Univ., Pittsburgh, PA)

Lip rounding enhances the acoustic contrast between /s/ and /ʃ/ in English. For speakers of languages without this phonemic distinction, to what degree does lip rounding affect the production and perception of these fricatives? Is visual awareness of lip rounding associated with audiovisual integration commonly observed in the McGurk effect? How does this performance develop as a function of exposure to English? Fourteen learners of English participated in three tasks. First, participants read a list of /s/-/ʃ/ minimal pairs. The spectral center of gravity was measured to examine learners' production of the contrast. Second, participants watched/listened to the same words produced by a native English speaker in four presentation formats: audio only, audiovisual-congruent, audiovisual-incongruent (e.g., an audio /s/ paired with a visual /ʃ/), and visual only. Finally, participants listened to four stop-vowel syllables paired with either congruent or incongruent videos. Two experimental sessions were conducted to examine the development of the performance. Preliminary results showed identification accuracy of /s/, compared to /ʃ/, was affected to a greater extent by presentation format. Response to /s/ was slower in the first session, but was comparable to /ʃ/ in the second session.

4aSC6. International teaching assistants' production of focus intonation. Sophia Kao (Linguist, Stony Brook Univ., SUNY at Stony Brook, Stony Brook, NY 11794-4376, sophia.kao@stonybrook.edu), Jiwon Hwang (Asian & Asian American Studies, Stony Brook Univ., Stony Brook, NY), Hyunah Baek, Chikako Takahashi, and Ellen Broselow (Linguist, Stony Brook Univ., Stony Brook, NY)

We report on a longitudinal investigation of the realization of English focus by 19 Mandarin-speaking International Teaching Assistants (ITAs). Participants read passages containing contrastive information (e.g., *The price of a train ticket is twenty dollars, while the price of a bus ticket is eleven dollars*), and then responded to the experimenter's questions (*Is the price of a bus ticket twenty dollars*?). ITAs were tested within a month of their arrival in the US, and again at the end of their first semester. Overall, the productions of ITAs at both points in time were judged as less natural by native English listeners than the productions of the native speakers of English, though the naturalness of some ITA productions improved at the second sampling. Acoustic analyses of the ITA productions and comparison with the productions of 18 native English speakers revealed a good deal of interspeaker variability in the ITA productions, with several different patterns associated with the "unnatural" productions: (a) failure to accent the focused element; (b) failure to deaccent the word following the focused element; and (c) failure to align the accent with the stressed syllable of the focused word, with the entire focused word spoken on high pitch.

4aSC7. Use of language-specific speech cues in highly proficient second-language listening. Anne Cutler (Univ. of Western Sydney, 1 Tewkesbury Ave., Apt. 60, Darlinghurst, NSW 2010, Australia, a.cutler@uws.edu.au), Laurence Bruggeman (Univ. of Western Sydney, Penrith, NSW, Australia), and Anita Wagner (UMCG, Groningen, Netherlands)

Language-specificity in listening to speech occurs at all processing levels and even between structurally close languages (e.g., English, Dutch). Transitional cues to fricative place of articulation are used in English for identifying /f/ (which resembles theta) but not /s/, whereas in Dutch (without theta) they are used for neither. In spoken-word recognition, suprasegmental cues are used in Dutch, but not in English (with more segmental reduction); Dutch L2 listeners even outperform native L1 listeners in detecting origin of differently stressed English syllables (e.g., car- from CARton versus carTOON). Here, longterm residents in Australia with Dutch as L1 but predominantly using English completed each of these tasks. In the phonetic task, with cross-spliced nonsense words, these listeners performed just as Dutch listeners in the Netherlands, showing insensitivity to transitional cues for both /f/ and /s/. In the lexical task, with word fragments (e.g., car-), they however did not behave as L1 Dutch and outperform Australian English listeners, but instead resembled the latter, by ignoring suprasegmental stress cues. A (lexical) listening strategy available in L1 can apparently be abandoned if it delivers little payoff in L2, but acquiring for L2 listening a (phonetic) strategy not used in L1 seems less feasible.

4aSC8. The perceptual assimilation model for suprasegmentals and cross-language lexical-tone identification. Jennifer Alexander (Dept. of Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, Jennifer_Alexander@northwestern.edu) and Dr. Yue Wang (Dept. of Linguist, Simon Fraser Univ., Burnaby, BC, Canada)

We examine how native lexical-tone experience influences identification of novel tone. Cantonese, Thai, Mandarin, and Yoruba listeners identified CV syllables bearing the six phonemic Cantonese tones. Accuracy scores were submitted to a two-way rANOVA with L1-Group (x4) as the between-subjects factor and Tone (x6) as the within-subjects factor. Tone error patterns were also assessed via rANOVAs with L1-Group (x4) as the between-subjects factor and Response-Pattern (% correct versus % other response) as the within-subjects factor. Consistent with previous reports, native listeners' confusions reflected effects of ongoing tonal mergers and a crowded tone space. Non-native listeners appeared to assimilate novel tones to L1 tone categories by attending to phonetic cues relevant to the phonological and phonetic properties of their L1s. Overall, results support predictions of the Perceptual Assimilation Model for Suprasegmentals (PAM-S). [Support: NSF grant 0965227 to J.A.A.]

4aSC9. The language-familiarity effect in talker identification by highly proficient bilinguals depends on second-language immersion. Sara C. Dougherty and Tyler K. Perrachione (Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, sarad12@bu.edu)

Listeners identify voices more accurately in their native language than an unfamiliar, foreign language in a phenomenon called thelanguage-familiarity effect (LFE). Talker identification studies consistently find that even highly proficient speakers of a foreign language persist in identifying voices more accurately in their native language than their second language. It remains unclear why some bilinguals continue to exhibit the LFE despite fluent receptive and expressive second-language skills. In the present study, native Mandarin speakers who were fluent in English and living in the United States learned to identify Mandarin- and English-speaking talkers by the sound of their voice. We assessed hypotheses proposing that either depth of second-language immersion or age of second-language acquisition account for the magnitude of the LFE among highly proficient second-language speakers. Only the extent to which bilinguals were currently immersed in second-language use affected the magnitude of the LFE; native Mandarin speakers who currently used predominately English in their daily lives exhibited no LFE, whereas those who currently used predominately Mandarin continued to have superior native-language talker identification abilities. These results suggest that the magnitude of the LFE in talker identification by bilinguals is a function of real-world immersion in foreign language use.

4aSC10. Vowel length and perception of English as a Lingua Franca. Mara Haslam and Elisabeth Zetterholm (Dept. of Lang. Education, Stockholm Univ., Stockholms Universitet, Institutionen för Språkdidaktik, Stockholm 10691, Sweden, mara.haslam@isd.su.se)

English is one of the most widely used languages in the world but, in contrast to many other languages, a majority of users of English are non-native speakers, meaning that many interactions in English fall under the category of English as a Lingua Franca (ELF). Understanding of how non-native speakers' perception works in English is important for both researchers and English teachers. Jenkins' (2000) Lingua Franca Core (LFC), a list of pronunciation characteristics which are claimed necessary for accurate ELF perception, has already been adopted as a standard for training English teachers and for students of English. However, the research on which the LFC is based is rather limited. An earlier study by the presenters about the importance of the fortis-lenis distinction on stop consonants for ELF perception indicates that participants' perception of word-initial stops cannot be explained easily in relation to voice onset time. This study represents a further attempt to investigate the claims of the LFC using more controlled methods, focusing on the claim that length is important for accurate vowel perception. Word tokens from conversations in English between speakers with different L1s are used. Results of vowel-length measurements and their relationship to perceptual results will be presented.

4aSC11. Visual and auditory native language interference in perceptual learning of non-native speech sounds. Pamela Fuhrmeister, F. Sayako Earle (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., U-1085, Storrs, CT 06269, pamela.fuhrmeister@uconn.edu), Jay Rueckl (Psychol. Sci., Univ. of Connecticut, Storrs, CT), and Emily Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, Storrs, CT)

Learning speech sounds in a second language is challenging for adults, especially when non-native phonemes are perceptually similar to those in the native language. Results from Earle and Myers (2015) suggest that auditory exposure to native language tokens that are perceptually similar to a learned non-native phonetic contrast attenuate sleep-mediated gains in perceptual learning. The present study seeks to determine whether activating abstract phonetic category representations through visual input will produce a similar interference effect on a learned non-native phonetic contrast. To test this, we trained participants to identify the Hindi dental and retroflex contrast and reassessed their performance following a period of sleep. Immediately after training, participants were exposed to interference tokens through a pseudohomophone judgment task, in which they were asked to decide if a string of letters spelled out a real word when read aloud (e.g., "drane" = drain). One group read words with /d/-initial tokens (perceptually similar to the learned contrast) and another read /b/-initial tokens (perceptually dissimilar). The effect of visual interference on perceptual performance differed from the effects observed with auditory interference. These results suggest that different modalities of language input differentially interfere with perceptual learning of non-native speech sounds.

4aSC12. The effect of visual information on non-native speakers' perception of Cantonese tones. Yan Chen (Linguist Dept., Univ. of Arizona, TUCSON, AZ 85721, yanchen@email.arizona.edu)

This study examines the effect of tone marks on the perception of 5 difficult Cantonese tone pairs that have high perceptual similarity: high-rising (25) versus mid-rising (23), mid-level (33) versus low-level (22/11), low-falling (21) versus low-rising (23), low-falling (21) versus low-level (22/11), and low-rising (23) versus low-level(22/11). Native speakers of American English and native speakers of Mandarin participated in a categorial AXB pre-test, a categorial AX training, and a categorial AXB post-test. Half of the subjects received iconic symbols (˨˥ for high-rising (25), ˧ for mid-level (33), ˨˩ for low-falling (21), ˨˧ for low-rising (23), and ˨ for low-level (22/11)) as feedback in the training (Auditory-Visual training group) while the other half did not (Auditory). Preliminary results showed that both language groups improved their discrimination of all the tone pairs and that Mandarin speakers received higher percent response than English speakers in the post-test. AV subjects and A subjects, regardless of language groups, did not differ significantly in percent correct response in the post-test. Reaction time data suggested that AV subjects responded faster on the pair high-rising (25) versus mid-rising (23) than A subjects, regardless of language groups, and that Mandarin speakers responded faster than English speakers in the post-test.

4aSC13. Intelligibility, fluency, and variability in non-native speech. Melissa M. Baese-Berk (Linguist, Univ. of Oregon, Oyer Ctr. B-7, East Lansing, MI 48824, mbaesebe@uoregon.edu), Tuuli H. Morrill (George Mason Univ., Fairfax, VA), and Ann R. Bradlow (Northwestern Univ., Evanston, IL)

Native and non-native speech differ in many ways, including overall speech rate, which tends to be substantially slower for non-native speakers (Guion *et al*., 2000). Recent work has suggested that non-native speech may be not only slower, but also more variable when non-natives are reading aloud (Baese-Berk and Morrill, 2015). Speaking rate also influences how listeners perceive non-native speech—slower readers are perceived as more accented and less comprehensible (Munro and Derwing, 1998). In the present study, we ask whether variability in speaking rate also has this effect on listeners. Specifically, we ask whether speaking rate variability is correlated with non-native speaker intelligibility and/or with judgments of fluency. In the present study, we ask listeners to transcribe sentences from speakers who show more or less variability in speaking rate across sentences in a paragraph-length reading passage. We also ask them to rate the fluency of the read sentences. Speech samples were taken from native Mandarin, Korean and English speakers' recordings of the English "North Wind and the Sun" passage in the ALLSSTAR collection of digital speech recordings (Bradlow *et al*., 2010). These results provide insight into the relationships between variability, intelligibility, and fluency in both native and non-native speech.

4aSC14. Variability and stability in native and non-native Japanese speech. Charlotte Vaughn, Melissa Baese-Berk (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403-1290, cvaughn@uoregon.edu), and Kaori Idemaru (East Asian Lang. and Literatures, Univ. of Oregon, Eugene, OR)

The speech of second language learners is often characterized as more variable than the speech of native speakers (e.g., Baese-Berk and Morrill, 2015; Jongman and Wade, 2007; Jongman *et al*., 2007; Rogers*et al*., 2012). However, among the relatively few studies which have examined this question directly, the majority have focused on: learners of English, learners from one L1 background, and have examined one linguistic feature per study, leaving open the possibility that non-native speech may not be more variable than native speech in all cases. The present work examines the production of Japanese sentences by native Japanese speakers, and learners of Japanese from two language backgrounds (English and Chinese). Counter to previous results, we find several cases where non-native speakers are *less* variable than native speakers, including on various measures of vowel spectrum variability, and voiced stop realization. Thus, the relationship between variability and language background appears to be more complex than previously thought. Accurately characterizing variability across native and non-native speech has implications for models of second language acquisition, and speech perception. We suggest that future work on variability in non-native speech should examine factors such as speakers' L1 and L2, and multiple linguistic features.

4aSC15. Discriminability of non-native tonal contours in low-pass filtered speech. Tuuli Morrill and Zhiyan Gao (George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, tmorrill@gmu.edu)

Native speakers of a language are able to easily detect non-native "accents"—usually a perceived accent is attributed to phonemic- and phonetic- level information. However, a growing body of work suggests that non-native speakers also exhibit differences from native speakers at the prosodic level. Differences are exhibited in the placement of lexical stress and phrasal accents, as well as the acoustic cues used in prominence production; prosodic differences in production appear to pattern in (native)-language-specific ways (Morrill, *ICPhS Proceedings,* 2015). In this experiment, we ask (1) whether differences in the realization of tonal contours are perceived by listeners, and (2) whether perceptual discriminability is patterning

according to speakers' native language. Participants listened to low-pass filtered phrases from the Stella passage recordings on the GMU Speech Accent Archive—produced by native speakers of English, Mandarin, Korean, Arabic, and Turkish—and judged whether speakers were saying the same thing. Results indicate that participants were less likely to rate two phrases as the same (even when they were) if they had been produced by speakers of different native languages. Patterns of discriminability across language pairings are attributed to differences in the number and timing of tonal events (e.g., pitch accents).

4aSC16. Recognition memory for foreign- and native-accented sentences. Kristin Van Engen (Washington Univ. in St. Louis, One Brookings Dr., Campus Box 1125, Saint Louis, MO 63130-4899, kvanengen@wustl.edu) and Jonathan Peelle (Washington Univ. in St. Louis, St. Louis, MO)

The acoustic-phonetic properties of speech signals affect not only their intelligibility, but also how well they are encoded in memory. Recognition memory can be improved, for example, when speakers intentionally speak clearly (Van Engen *et al.*, 2012). One explanation for this result is that enhanced acoustic-phonetic cues reduce the cognitive effort associated with perceptual speech processing, thereby increasing the availability of processing resources for encoding speech content in memory. In the current study, we tested the hypothesis that, following the same logic, recognition memory would be reduced for foreign-accented speech, in which acoustic-phonetic cues deviate from native-language norms. Participants heard English sentences produced by a native speaker of English and a native speaker of Korean. Recognition memory was tested using an old-new judgment task in which participants heard those recordings again, along with an equal number of new items. Surprisingly, recognition memory was higher for Korean-accented English than for native-accented English. A follow-up study in which test sentences were presented visually showed no difference between foreign- and native-accented speech. These results are consistent with an acoustic distinctiveness account in which foreign-accented speech can in some cases be remembered more accurately due to its distinct acoustic features.

4aSC17. Categorization training for non-native accented word recognition. Rachel Tessmer (Commun. Sci. and Disord., The Univ. of Texas at Austin, 2504 Whitis Ave., A1100, Austin, TX 78712, tessmer.rachel@gmail.com), Eriko Atagi (Volen National Ctr. for Complex Systems, Brandeis Univ., Waltham, MA), Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN), and Bharath Chandrasekaran (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

Non-native accented speech is considered an adverse listening condition for native speakers (Mattys *et al.*, 2012). Previously, accurate categorization of non-native accents was found to relate to accented speech-in-noise recognition (Atagi and Bent, 2015). The goal of the current study is to examine (1) the extent to which native speakers can be *trained* to categorize the language backgrounds of non-native talkers using feedback, and (2) the extent to which accent categorization training, relative to word and sex identification tasks, improves non-native accented word recognition. Participants were randomly assigned to one of three training tasks: an accent categorization task with trial-by-trial feedback ($n=10$), a transcription task ($n=10$), and a speaker sex categorization task ($n=10$). Critically, all participants were exposed to the same set of training stimuli. Participants completed a transcription task with speech-shaped noise prior to and following training. We assessed keyword accuracy for non-native accented speech and native accented speech. Our results demonstrate that single-session training can improve accent categorization. Accent identification training, word identification, and sex identification all yielded more accurate non-native speech perception (post>pre). We find that irrespective of training task, exposure to non-native accents enhances intelligibility of non-native accented speech.

4aSC18. The bilingual advantage in phonetic/phonological learning: A study of bilingual and monolingual patterns in the reproduction of word-final stops in three novel accents of English. Laura Spinu (Anthropology, Univ. of Western ON, Western University, London, ON N6A 3K7, Canada, lspinu@uwo.ca), Jiwon Hwang (Asian & Asian American Studies, Univ. of Stony Brook, Stony Brook, NY), and Renata Lohmann (Univ. of Western ON, London, ON, Canada)

As part of a larger study investigating the acoustic correlates of accentedness in the reproduction of various accents of English by English monolinguals and French-English bilinguals, we explored speakers' ability to imitate and spontaneously reproduce patterns of realization of word-final coronal stops in three different accents: SE England (Sussex) in which these stops are 100% glottalized, and Russian English and South African English, in which these stops exhibit canonical release about 50% of the time. We have so far fully analyzed the Sussex results and partially analyzed the Russian results. The two groups were characterized by different behaviors: while the bilinguals successfully reproduced the Sussex English accent, the monolinguals did not. By contrast, neither of the groups was successful in reproducing the Russian English accent. After considering the characteristics of each group of speakers and each accent, we conclude tentatively that it is the bilinguals, as a group, who were more successful in the phonetic/phonological learning of a new pattern, perhaps as a result of some type of "bilingual advantage." Based on work by Calabrese (2011) and Krizman *et al.* (2012), we propose that this advantage stems from longer availability of acoustic information in echoic memory.

4aSC19. Transitioning learning strategies in speech categorization enhances lexical learning. Han-Gyol Yi, Rachel Tessmer, and Bharath Chandrasekaran (Commun. Sci. and Disord., The Univ. of Texas at Austin, 2504 Whitis Ave., A1100, Austin, TX 78712, tessmer.rachel@gmail.com)

A recent model suggests two strategies are involved in speech category learning: a reflective strategy that maps sounds onto categorical representations explicitly, and a reflexive strategy that does so implicitly. Successful learners transition from an initial reflective strategy to a reflexive strategy. We examined the extent to which speech category learning in adults can be improved by selectively enhancing these strategies in an optimal sequence, and how this influences performance in a lexical learning task. Monolingual English speakers ($N=40$) learned to categorize Mandarin tones. Participants in the "optimal" condition ($n=10$) were first presented with a combination of feedback information and talker presentation designed to enhance reflective strategies, and later presented with a feedback-talker combination that targeted reflexive strategies. This order was reversed in the "sub-optimal" condition ($n=10$). Two control conditions ($n=10$ each) exclusively targeted either strategy. All participants then learned 24 pseudo-Mandarin lexical items across three days. Participants in the optimal condition outperformed those in the sub-optimal and control conditions on the lexical learning task despite the fact that they did not attain the highest speech categorization accuracy. These results suggest that transitioning from reflective to reflexive strategies is ideal for applying learned speech categories to novel words.

4aSC20. Non-native speakers' acoustic variability in producing American English tense and lax vowels. Bruce L. Smith (Commun. Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Rm. 1201, Salt Lake City, UT 84112, bruce.smith@hsc.utah.edu) and Rachel Hayes-Harb (Linguist, Univ. of Utah, Salt Lake City, UT)

The primary goal of the present study was to compare native (L1) and non-native (L2) speakers' vowel productions to determine whether the L2 subjects showed similar or greater within-subject (i.e., token-to-token) variability relative to the L1 talkers when producing American English vowels. We were particularly interested in whether L2 subjects whose vowels were

more native-like would be less variable in their productions and whether L2 talkers' whose vowels were less native-like would be more variable in their productions. First and second formants of the tense and lax vowel pairs / i—ɪ /, / e—ɛ /, and / u—ʊ / were measured and Coefficient of Variation was calculated for 10 native speakers of American English and 30 non-native speakers of English from three different language backgrounds (viz., 10 Mandarin speakers, 10 Korean speakers, and 10 Spanish speakers). Overall, the L2 subjects' vowel formant patterns were found to be comparable to those of L1 speakers approximately half of the time. Whether the L2 talkers' American English vowel formants were native-like or not, however, they seldom showed greater within-subject variability than the native speakers.

4aSC21. The effect of speaking style adaptations on speech perception in noise by native and non-native listeners. Kirsten Meemann (English, Univ. of Texas at Austin, 204 W. 21st St., Mail Code B5000, Austin, TX 78712-1164, kirsten.meemann@utexas.edu) and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

It is well established that non-native listeners perform worse on speech perception in noise tasks compared to native listeners. However, few studies have directly compared the effect of speaking style adaptations on speech perception in energetic and informational maskers for these two listener groups. The present study examined intelligibility of conversational (CO), clear (CS) and noise-adapted (NAS) meaningful sentences mixed with speech shaped noise (SSN) and 2-talker (2T) babble. Sentences were presented at −5 dB signal to noise ratio (SNR) to native and −3 dB SNR to non-native listeners. The results revealed that CS and NAS significantly improved word recognition in noise for both listener groups. However, the gains in intelligibility were substantially greater for native compared to non-native listeners. Word recognition was overall better in 2T babble than in SSN. Our findings confirm that both native and non-native listeners benefited from speaking style adjustments, but native listeners were better able to utilize the intelligibility-enhancing modifications. The results also suggest that informational masking in 2T babble is less disruptive than energetic masking in SSN. This may in part be due to the "glimpsing" windows through which the target speech could become more easily accessible.

4aSC22. Adaptation to accent is proportionate to the prevalence of accented speech. Matt Lehet and Lori L. Holt (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Carnegie Mellon, Baker Hall-346, Pittsburgh, PA 15213, mil@andrew.cmu.edu)

Listeners rapidly reweight the mapping of acoustic cues to speech categories in response to abrupt introductions of accented speech. For instance, when encountering an accent that reverses the typical correlation of acoustic cues to speech category membership, listeners rapidly down-weight reliance on secondary cues [Idemaru and Holt, 2011; 2014; Liu and Holt 2015]. Here, we examined the impact of experiencing mixtures of typical and accented speech in evoking cue down-weighting to test how much information the system requires to adapt. Listeners recognized words as *beer* versus *pier* across 41 blocks of 20 trials. In the first block, participants exclusively encountered tokens with the canonical English relationship between fundamental frequency (F0) and voice onset time (VOT); low F0 frequencies were associated with short VOT durations whereas high F0 frequencies were associated with long VOT durations. In subsequent blocks, the ratio of canonical to accented tokens (for which the F0/VOT correlation was reversed) was changed by 5% per block until participants experienced exclusively accented speech. Canonical tokens were then reintroduced incrementally, until participants exclusively heard canonical speech. Reliance on F0 for word recognition linearly decreased as the proportion of accented speech increased. Implications for models of the speech perception system are discussed.

Session 4aSP

Signal Processing in Acoustics and Underwater Acoustics: Detection and Estimation in Uncertain Acoustic Environments I

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

Chair's Introduction—10:30

Invited Papers

10:35

4aSP1. Information-theoretic analysis of Bayesian localization of a narrowband source in an uncertain environment. Thomas J. Hayward (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Previous work has explored the application of fundamental information-theoretic constructs, including posterior entropy of source location and per-iteration information gain (relative entropy) and their large-ensemble limits, to quantify the performance of iterated (sequential) Bayesian localization of an acoustic source. These information-theoretic quantities are closely tied to Bayesian inference and provide global measures of the uncertainty of source location that is represented by the Bayesian posterior probability density. The present work extends this analysis to the localization of a narrowband acoustic source in an acoustic environment having uncertain sound speed and attenuation represented by a joint pdf of the two quantities. The fundamental principle relating the environmental and acoustic field uncertainties is the change-of-variables theorem of probability theory. The degradation of localization performance due to the environmental uncertainty is quantified by the increase in posterior entropy of source location, interpreted as an information loss. Examples are presented, and extensions to stochastic environments are discussed. [Work supported by ONR.]

10:55

4aSP2. Removing multiple scattering in underwater target detection. Katherine F. Woolfe, Douglas Photiadis, and David Calvo (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC, katherine.woolfe.ctr@nrl.navy.mil)

Recent advances have been made in the field of optics to image through multiple-scattering media in a regime where classical imaging techniques, which rely on the single-scattering approximation, fail. Using results generated by random matrix theory, it is possible to filter the response matrix generated by a source/receiver array to remove the multiple scattering components of the received signals. After removing the multiple scattering components, classical imaging techniques can be used to image with the single-scattering components. We show how this single-scattering filter can be adapted for underwater acoustical imaging in cases where multiple scattering effects are large compared to the reflection from the target. A full-wave 3-dimensional time-domain scaled model is used to characterize filter performance as a function of target strength and environmental coherence length for a shallow-water case. Preliminary results indicate that the filter can be used to detect a target located at least 2 mean free paths away from the source/receiver array.

Contributed Papers

11:15

4aSP3. Additional studies of the acoustics of coffee roasting. Jay R. Johnson and Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, 1 University Station C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu)

Cracking sounds emitted by coffee beans during the roasting process can be recorded by a microphone and used as the basis for automated acoustic roast profiling and monitoring, mimicking what expert artisanal coffee roasters do by ear. Three parameters are used for this purpose. Near the end of the roasting process, sounds known as "first crack" exhibit a higher acoustic amplitude than sounds emitted later, known as "second crack." First crack emits more low frequency energy than second crack. Finally, the rate of cracks appearing in the second crack chorus is higher than the rate in the first crack chorus. This presentation is a companion to the previously published work on the same topic [J. Acoust. Soc. **185**, EL265–EL269 (2014)], but expanded to include a discussion of automated crack detection signal processing, the acoustic characteristics of different coffee beans, and initial results of a study on how individual beans crack and emit their sounds.

11:30

4aSP4. Learning a linear ordered-statistic constant false-alarm rate detector. Timothy C. Havens (Elec. and Comput. Eng., Michigan Technolog. Univ., Houghton, MI), Ian Cummings, Jonathan Botts (Appl. Res. in Acoust. LLC, Culpeper, VA), and Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

The linear ordered statistic (LOS) is a parametrized ordered statistic (OS) that is a weighted average of a rank-ordered sample. LOS operators are useful generalizations of aggregation as they can represent any aggregation, from minimum to maximum, including conventional aggregations,

such as mean and median. Hence, we propose an LOS-Constant False-Alarm Rate (CFAR) detector that uses an LOS operator to compute the background level. Specifically, we extend ordered weighted average (OWA) learning methods to learn an LOS-CFAR using two regularization methods, L2 and entropy regularization. We demonstrate our learning process for midfrequency active sonar in cluttered shallow-water environments using both synthetic data and physics-based simulated experiments, and present and discuss the performance of the learned CFAR detectors versus the classical OS-CFAR and cell-average (CA)-CFAR. [Work supported by a NAVSEA Phase I SBIR award.]

11:45

4aSP5. Extrapolation of measured correlation replica fields in passive acoustic source localization. Christopher M. Verlinden (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu), J Sarkar (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), K G. Sabra (Mech. Eng. Dept., Georgia Inst. of Technol., Atlanta, GA), and W A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

A method of localizing unknown acoustic sources using data derived replicas from ships of opportunity has been previously reported (Verlinden, 2015). The method is capable of localizing sources in positions where data derived replicas are available, such as locations previously transited by ships, tracked using the Automatic Identification System (AIS). Here, we present an extension of this localization method to regions where data derived replicas are not available by extrapolating the measured replicas onto a more extensive grid. This new augmentation provides the additional opportunity for continuous tracking.

THURSDAY MORNING, 26 MAY 2016 — SALON A, 8:00 A.M. TO 11:10 A.M.

Session 4aUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Sediment Characterization Using Direct and Inverse Techniques III

David P. Knobles, Cochair
KSA LLC, PO Box 27200, Austin, TX 78755

Preston S. Wilson, Cochair
Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292

Chair's Introduction—8:00

Invited Papers

8:05

4aUW1. Seabed property inversion—Sequential and direct approaches. Zoi-Heleni Michalopoulou, Tao Lin (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu), and Nattapol Aunsri (School of Information Technol., Mae Fah Luang Univ., Tasud, Muang District, Chiang Rai Province, Thailand)

The goal of this work is to improve our knowledge of the ocean medium using physics of sound propagation and statistical signal processing. Two of the approaches we discuss are based on global and local optimization combined with sequential filtering. A third one falls under the category of direct ("exact") methods. The first two approaches are based on the extraction and the association between paths/modes and detected arrivals and also produce posterior probability density functions, characterizing the uncertainty structure of modal/multipath arrivals. These are then propagated backwards through an acoustic model for posterior probability density function calculation of environmental parameters. A third method extends Stickler's exact inverse approach, employing measurements of a reflection coefficient at low frequencies to obtain the sediment sound speed profile. We develop an approximation based on interpolation and a regularization approach. The former improves on results of a previous method for solving the trace equation using a linear approximation. The latter assists in addressing singularities caused by noise in the data. Several assumptions are initially made in order for the approach to work. We show via a sensitivity analysis that the method is robust with respect to initial assumptions and parameter choices needed for both the interpolation and regularization processes. [Work supported by ONR.]

8:25

4aUW2. Modal wavenumber estimation using a Bayesian compressed sensing algorithm. Angelique Dremeau, Julien Bonnel, and Florent Le Courtois (Lab-STICC, ENSTA Bretagne, 2 rue François Verny, Brest cedex 9 29806, France, julien.bonnel@ensta-bretagne.fr)

In shallow water, low-frequency acoustic propagation is described by modal theory. In this context, the environment acts as a dispersive waveguide, and the geoacoustic properties of the seabed can be inferred from the modal wavenumbers. When considering horizontal aperture (using a horizontal array or a towed source), wavenumber estimation is a well-known problem, equivalent to spectral analysis. In this paper, wavenumber estimation is revisited using Compressed Sensing (CS). Our method benefits from two strong physical hypotheses. Only a few modes are propagating so that the wavenumber spectrum is sparse. Moreover, if the source is broadband, the wavenumbers can be related from one frequency to the next using a general dispersion relationship. Our method resorts to a state-of-the-art Bayesian algorithm exploiting a Bernoulli-Gaussian model. The latter, well-suited to the sparse representations, makes possible a natural integration of the prior dispersion information through a wise choice of the Bernoulli parameters. The whole methodology is assessed on simulated data and successfully applied on marine data.

8:45

4aUW3. Maximum entropy estimation of environmental parameter distributions. Richard Pitre (RobTre Res., L.L.C., 58 Crystal Canyon Dr., Carbondale, CO 81623, richard.pitre@robtre.com)

This presentation regards origins and information theoretic foundations of maximum-entropy methods for estimating environmental parameter distributions. Environmental parameter distributions determine uncertainty and confidence levels in transmission loss inputs to mission planning and sonar tactical decision making tools. Given the scale of the oceans and their stochastic-dynamic nature, the sample base for estimation is usually limited relative to the required spatial and temporal resolution. Effective use of estimated environmental distributions for planning and decision making in this context requires well defined and meaningful methods of extrapolative estimation. The principle of maximum-entropy or minimal-presumption provides an important component of such methods. A complete definition of a maximum entropy method requires (1) definition of an application specific information or information base measure that counts the relative number of physically distinct possibilities in measurable subsets of the state space, and (2) a way of imposing minimally informative extrapolation on a combination of data samples, statistical hypotheses, and mathematical presumptions regarding things like smoothness of the underlying distribution. These components of distribution estimation and their interrelationships are discussed.

9:05

4aUW4. Compressive acoustic sound speed profile estimation using wavelets. Michael J. Bianco, Haiqiang Niu, and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr, La Jolla, CA 92037, mbianco@ucsd.edu)

Estimation of ocean acoustic sound speed profiles (SSPs) requires inversion of acoustic data with limited observations. Inversion for true ocean SSP structure is a nonlinear, underdetermined problem that requires regularization to ensure physically realistic solutions. Traditional regularization, which minimizes the energy of best-fit solutions, requires undersampling of true SSPs or using few shape functions. This gives low resolution SSP estimates which can affect the accuracy of other parameters in geoacoustic inversion. Compressive sensing (CS) reliably estimates signal parameters for certain highly underdetermined linear problems provided the signal can be "compressed:" represented in a sparse domain where few non-zero coefficients (out of many) explain the observations. Here, it is shown that ocean SSPs can be compressed using dictionaries of wavelets, empirical orthogonal functions, and other shape functions given *a priori* environmental statistics. Shape dictionaries are sought which represent SSPs using the fewest coefficients. These optimal dictionaries are used with CS to estimate SSPs by inverting a linearized normal mode model. It is shown that range-independent ocean SSPs are estimated robustly and in high resolution using CS.

Contributed Papers

9:25

4aUW5. Representation of depth-dependent gradients in sediment geoacoustics by Bernstein polynomials. Jorge E. Quijano, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. (Ring Road), Victoria, BC V8P 5C2, Canada, sdosso@uvic.ca), Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA), and Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

We present a seabed parametrization approach for depth-dependent gradients in sediment geoacoustics, a property commonly observed in muds. The method represents continuous functions by a polynomial form, consisting of a finite sum of Bernstein basis weighted by real coefficients which are estimated by Bayesian geoacoustic inversion of seabed reflectivity data. The advantages of the Bernstein representation of continuous gradients are discussed, including efficiency in representing a wide variety of gradients with only a few coefficients, as well as high numerical stability of the polynomial form to perturbation of its coefficients. The performance of the Bernstein parametrization applied to geoacoustic inversion is illustrated with simulated data obtained from a realistic seabed scenario. In addition, the Bernstein approach is applied to experimental data from four mud sites at the Malta Plateau. The estimated geoacoustic profiles are in good agreement to core measurements from the area, and serve to illustrate the ability of the Bernstein-based inversion to represent steep gradients. Comparison to results obtained by discrete (multi-layered) and other continuous gradient representations (line-, sinusoid-, and spline-based) is presented.

9:40

4aUW6. On connection between statistical integral equations and statistical inference in ocean acoustics and sonar. David P. Knobles (Knobles Sci. and Anal., LLC, PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com) and Preston S. Wilson (ARL, Univ. of Texas, Austin, TX)

Under what circumstances do statistics of source and waveguide parameter values derived from an application of statistical inference, such as Bayesian or Maximum Entropy, reflect the temporal and spatial inhomogeneities of the waveguide? The statistics of the objective function is influenced by model error that reflects the non-deterministic features of the

waveguide and the source. This connectivity is explored with a new derivation of statistical equations previously discussed by Dozier and Tappert. The resulting statistical integral equations provide the relationship between the fluctuations in the waveguide to those of the acoustic field. Then, the idea is to infer, from an ensemble of measured acoustic data, marginal probability distributions for the Green's functions and mode coupling matrix coefficients. This is accomplished by inferring the statistics of the source and the physical properties of an ocean waveguide such as range-dependent sound speed layers in the water column and the seabed, and layer interface roughness parameters. The inferred statistics of the mode coupling operator and Green's functions, and the noise spatial correlation then become input into the statistical integral equations. In principle these modeled acoustic field statistics can then be compared to the measured field statistics. [Work supported by ONR Code 32 OA.]

9:55–10:10 Break

10:10

4aUW7. A new time-domain model for bottom backscatter. Dajun Tang and Darrell R. Jackson (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Bottom backscatter is important for a number of underwater applications: it is a source of noise in target detection and a source of information for sediment classification and geoacoustic inversion. Bottom backscatter measures sound scattered from roughness and volume heterogeneity in different sediment layers. While current models can successfully handle scattering from such sediment layers, they cannot correctly predict the scattered intensity as a function of time. A new model for bottom backscatter is introduced which provides an exact solution for scattered intensity as a function of time under first order perturbation theory. Examples of backscatter from various sediment layers will be shown in contrast to previous models. Issues such as the "spherical wave effect" will be addressed, and applications to model shallow water reverberation will also be discussed.

10:25

4aUW8. Spatial variability of seabed properties on the Malta Plateau inferred with an autonomous underwater vehicle. Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3P6, Canada, jand@uvic.ca), Charles W. Holland (Appl. Res. Lab., Penn State Univ., State College, PA), Stan E. Dosso, Jorge E. Quijano, and Eric Mandolesi (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

We develop an automated method to infer geoacoustic properties along tracks surveyed by autonomous underwater vehicles (AUVs). The AUV tows a 32-element receiver array and a source emitting signals ("pings") at regular intervals. Recordings are processed in terms of reflection coefficients, resulting in large data volumes with substantively more information on seabed structure than traditional seismic profiling. However, interpreting seabed spatial variability requires efficient inversion. The inverse problem is non-linear and requires Bayesian sampling to quantify parameter uncertainties. To account for changes in the number of seabed layers at each ping position, the parametrization treats this number as unknown with a Poisson prior and even-numbered order statistics to improve efficiency. The method is applied to 340 data sets along a 14-km track on the Malta Plateau, employing 8 graphics processing units for approximately 2 weeks of computing time. The results resolve layering along the track with previously unreported detail. An erosional boundary is clearly resolved as a high-velocity, high-density layer and appears rougher and is buried deeper in shallower water. Depressions along this boundary are filled in with lower-velocity material. In addition, sound attenuation is well constrained in a thick low-velocity wedge. [Work supported by ONR and SERDP.]

10:40

4aUW9. Estimation of geoacoustic model parameters from modal amplitude information. N. Ross Chapman (School Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada, chapman@uvic.ca) and Rui Duan (Inst. of Acoust. Eng., Northwestern Poly-Tech. Univ., Xi'an, China)

This paper analyzes an approach for estimating geoacoustic model parameters from information contained in normal modes of a broadband signal. Propagating modes are resolved by time-warping deconvolved signals from light bulb sound sources deployed at short ranges in shallow water. Amplitudes of the resolved modes contain information about sediment sound attenuation through the modal attenuation coefficient. However, the coefficient also depends on the sediment sound speed and density. A sequential inversion approach was developed that enables effective use of modal amplitudes to estimate sound attenuation in sediments. The inversion is based on a sequential Bayesian approach applied to features of resolved modal data that are highly sensitive to specific geoacoustic model parameters. Travel times of modal frequency components are inverted first for sediment sound speed and sediment layer thickness, and these estimates are used in subsequent stages. The effects of errors in estimates from previous stages are analyzed for the impact on estimates of sound attenuation in the final stage. In particular, it is shown that the sediment density is weakly sensitive and does not have significant impact on the estimation of attenuation.

10:55

4aUW10. Effects of seabed curvature on the scattered acoustic field. Sheri Martinelli and Charles Holland (Marine & Physical Acoust., ARL Penn State, P.O. Box 30, Mailstop 3230D, State College, PA 16804-0030, slm77@psu.edu)

Knowledge of the acoustic properties of the sea bottom provides valuable input into the development and performance of information processing algorithms designed to detect and identify, e.g., man-made objects near to, or embedded in, the seabed. This information is also useful for improving modeling and simulation results in reverberation-limited environments. A critical challenge in this area is in allowing for range dependent bathymetry as it can affect the propagating acoustic field in ways that are difficult to predict. Typical approaches to modeling variable bathymetry assume either piecewise flat or tilted (linear) geometry; however, this only serves as a discrete approximation to reality. This study investigates the effects of mild curvature on the 3D acoustic field in frequency and angle space via finite difference time domain simulations. We present results and discuss the implications on forward modeling of the acoustic interaction with the seabed. [Work supported by the Strategic Environmental Research and Development Program (SERDP).]

Session 4pAA

Architectural Acoustics and Musical Acoustics: Opera Rehearsal and Performance Spaces

Robin S. Glosemeyer Petrone, Chair
Threshold Acoustics.com, 53 W Jackson Blvd., Suite 815, Chicago, IL 60604

Chair's Introduction—1:00

Invited Papers

1:05

4pAA1. Looking to history for a modern paradigm in opera house design. Robin Glosemeyer Petrone and Scott D. Pfeiffer (Threshold Acoust. LLC, 141 West Jackson Blvd., Ste. 2080, Chicago, IL 60604, robin@thresholdacoustics.com)

The design of a 600-seat opera house is a rare opportunity. The cost of modern opera production leads to much larger facilities and small venues are more frequently multi-purpose venues by nature, leading to the use of concert hall or lyric theatre forms that can be pressed into service for opera. The United States has many small theatres, mostly on college campuses, but few see frequent use for Opera and as a result design choices tilt away from optimal for Opera performance. Historic opera houses of Europe present more opportunities for exploration. Many follow the Italianate form which is intimate and evokes our sense of what an opera house "should" be, though closer investigation exposes the weakness in these designs both theatrically and acoustically. The experience for many patrons is removed and lonely with severely challenged sightlines and an acoustic experience that is frontal, dry, and unengaging. Our exploration took us to the Royal Opera in the Palace of Versailles whose form varies in key ways from the typical. While the construction techniques of Versailles hinder its acoustic response, as inspiration, a new form for intimate opera is created.

1:35

4pAA2. Navigating the design of a 600-seat opera house. Scott D. Pfeiffer and Robin Glosemeyer Petrone (Threshold Acoust. LLC, 53 West Jackson Blvd., Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

Negotiation of a design for a 600-seat opera and chamber music hall presents challenges of dimension. Can the main floor audience enter under the parterre? What is the relationship of the singer to the closest audience members, leaving room for a full-size orchestra? Two full concept designs were carried to fully plausible outcomes to test these questions as well as the question of the resulting acoustic environment in order to arrive at decisions amongst the design team and user groups. The power of auralization and visualization tools was brought to bear on the decision making process, supplementing comprehensive benchmarking studies. While each constituent went into the process with moderately to significant biases, all entered the final discussions having arrived with an open mind about the optimal outcome for the unique circumstances of the project. Communication among the team using the medium of greatest familiarity for multiple constituents led to an outcome that has broad support from the entirety of the design team.

2:05

4pAA3. The golden throat—Effect of proscenium zone shaping in opera houses on balance between stage and pit. Joseph W. Myers (Kirkegaard Assoc., 801 W. Adams St. 8th Fl., Chicago, IL 60607, jmyers@kirkegaard.com)

This paper investigates the importance of the walls and ceiling in the "throat" of an opera house—the zone surrounding the proscenium—for encouraging a balance that favors onstage singers over musicians in the orchestra pit. By looking at a variety of opera houses, the author tests the idea that surfaces in this zone that are solid and not strongly angled are especially beneficial for encouraging balance.

2:35

4pAA4. Opera acoustics in multi-use performing arts centers. Mark Holden (Jaffe Holden Acoust., 114A Washington St., Norwalk, CT 06854, MHolden@JaffeHolden.com)

There is very little information available on practical solutions for designing acoustics for an orchestra pit. This paper guides the reader through a brief history of orchestra pits and how to design them for the modern multi-use halls used by opera companies. Musicians generally abhor playing in a pit. They feel it is crowded, hot, loud, and a horrible environment in which to hear themselves and other musicians. The acoustician's goal is to improve the acoustic environment within the pit and find the optimal projection of music to the audience and stage. Drawings and details along with photos from pits designed for halls around the world will be presented from Mark Holden's new book Acoustic of Multi-Use Performing Arts Centers published by CRC Press in January.

3:05–3:20 Break

3:20

4pAA5. Improving listening conditions in partially covered opera pits or "The strings are working so hard…why can't I hear them?". Christopher N. Blair and Paul H. Scarbrough (Akustiks, LLC, 93 North Main St., Norwalk, CT 06854, pscarbrough@akustiks.com)

Since the introduction of the gargantuan orchestra to the world of opera by Wagner, opera house designers have been faced with the question of whether to place the orchestra in an open pit, a covered pit, or a partially covered pit. The open pit has long been favored by musicians but has it operational and economical drawbacks, including often large distances from conductor to singer, challenging pit/stage balance conditions, and reduced audience seating close to the stage. A fully covered pit works well for the special works that the Bayreuth master intended for this condition, but is inappropriate for the rest of the core operatic repertoire. The partially covered pit where some upstage musicians play under the stage is conceptually advantageous for a number of reasons, but can present challenges in uneven listening conditions within the orchestra (excessive loudness, ensemble difficulties) and in the house. In some instances, musician complaints have resulted in regulatory action. This paper explores the root causes of ensemble issues commonly found in partially-covered orchestra pits and presents specific solutions drawn from the consulting (and conducting) experience of the authors.

Contributed Papers

3:50

4pAA6. Design of Gran Teatro Nacional in Lima—A world class opera house in South America. José A. Nepomuceno and Julio Gaspar (Acústica & Sônica, Rua Fradique Coutinho, 955 cjt 12, São Paulo, São Paulo 05433-000, Brazil, info@acusticaesonica.com.br)

The design of Gran Teatro Nacional in Lima, Peru, started in 2009 as very special task. Peru needed a venue with world-class standards to host local and international events of opera, concerts, and ballet. The theater has a built area of about 26000 m^2 with 1500 seats capacity. It has a fully equipped fly tower with motorized winches, an 11 m tall acoustical shell, 4 double deck stage lifts, 2 orchestra lifts, and a pit for 100 players. The equipment complies with the most restrictive European standards. The acoustics oriented the design, from room shaping to its volume, from materiality to seats selection. The acoustical consultants assumed modified horseshoe plan to improved intimacy and three tier levels. The hall has two reverberation chambers and several acoustical banners to provide variability in the acoustical response of the hall. Computer modeling and physical model supported the design. Since the Gala Opening in 2012 the theater has been getting remarkable reviews from conductors, international orchestras, and artists in general. The theater complied with the mission of bringing a world class performing art center to Peru. The paper provides the history of the project, architectural, and construction details and acoustical data.

4:05

4pAA7. Improving orchestra and singer communication at the Brazilian Opera. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com)

Large opera stages present a challenge in maintaining optimum synchronization for principals or a chorus that are far upstage of the orchestra. Further complications arise when the orchestra located in a deep orchestra pit (typical) and only the conductor is visible. This condition induces disparity between the visual timing from the conductor and the arrival of sound from the orchestra. This paper describes how the implementation of electronic architecture solved the problem and improved acoustical conditions on the stage. In addition, the system was extended to the house to provide optimum conditions for symphony performances.

4:20

4pAA8. Analysis and mitigation design of acoustics for performance hall at the Catholic University. Jonathan P. Coyle, Christopher J. Kramer, Nicole A. Bull, Jacob Maclin, and Joseph Vignola (School of Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC, DC 20064, 36coyle@cua.edu)

This presentation will discuss a student level assessment designed with the goal to analyze a Music Hall located on the campus of The Catholic University of America. Ward Recital Hall is 97,712 ft^3, seating 108 and was built as a hall intended for liturgical musical performance, but since the early 1900's the purpose of the room has changed, since then minor improvements have been implemented. We have found that the three significant problems within the room are the HVAC noise contamination, stage projection homogeneity, and sound transmission through barriers (doors and windows). We will present the findings on each of the problems above in how the stage projection is not currently homogeneous, the issue with exterior noise contamination, and how the HVAC can directly affect the listening of viewers. We also will present remedies for each issue as well as the associated cost for the remedies. We will have a list for these problems with several different solutions, with the intent of having a good, better, and best solution with costs with respect to predictions of performance.

Session 4pAO

Acoustical Oceanography and Animal Bioacoustics: Acoustical Oceanographic Tools for the Study of Marine Ecosystems

David R. Barclay, Cochair
Department of Oceanography, Dalhousie University, PO Box 15000, Halifax, NS B3H 4R2, Canada

Wu-Jung Lee, Cochair
Psychological and Brain Sciences, Johns Hopkins University, 3400 N Charles St., Ames Hall 132, Baltimore, MD 21218

Chair's Introduction—1:00

Invited Papers

1:05

4pAO1. Relative impact of signal-to-noise ratio and propagation effects on the performance of an aural classifier. Carolyn Binder (Oceanogr. Dept., Dalhousie Univ., LSC Ocean Wing, 1355 Oxford St., PO Box 15000, Halifax, NS B3H 4R2, Canada, carolyn.binder@dal.ca) and Paul C. Hines (Dept. of Elec. and Comput. Eng, Dalhousie Univ., Halifax, NS, Canada)

Passive acoustic monitoring (PAM) is used to study marine mammals in their habitats, which cover diverse underwater environments. The distinct propagation characteristics of different ocean environments alters the time-frequency characteristics of a recorded signal. This may affect the accuracy of PAM systems. To develop a PAM system capable of operating under numerous environmental conditions, one must account for the impact of propagation. An aural classifier developed at Defence R&D Canada (DRDC) has successfully been used for inter-species discrimination of cetaceans. The aural classifier achieves accurate results by using perceptual signal features that model the features employed by the human auditory system. The current work examines the relative impacts of signal-to-noise ratio (SNR) and propagation effects on the performance of the aural classifier. DRDC's pulse propagation model, Waveform Transmission Through a Channel (WATTCH), was used to simulate signals travelling through the ocean environment over ranges of 0–20 km. Noise was added to both these signals and the original signals, so that performance could be compared for three scenarios expected to decrease classifier performance: decreasing SNR, increasing propagation effects (frequency spreading, multipath, etc.), and combined SNR and propagation effects. In this presentation, the modeled results are compared to experimental data.

1:20

4pAO2. Bowhead whale localization and environmental inversion using asynchronous hydrophones. Graham A. Warner, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. Ste. A405, Victoria, BC V8P 5C2, Canada, gwarner@uvic.ca), David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada), and Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper estimates bowhead whale locations and environmental properties using Bayesian inversion of the modal dispersion of whale calls recorded on asynchronous recorders in the shallow waters of the northeastern Chukchi Sea, Alaska. Bowhead calls were recorded on a Y-shaped cluster of seven autonomous ocean-bottom hydrophones, separated by up to 9.2 km. We use a warping time-frequency analysis to obtain frequency-dependent relative mode arrival times for nine frequency-modulated whale calls. Trans-dimensional inversion is applied to invert mode arrival times for the whale location, water sound-speed profile, subbottom layering and geoacoustic parameters, source instantaneous frequency (IF), relative recorder clock drifts, and residual error standard deviation, all with estimated uncertainties. Joint inversion of multiple calls is found to substantially reduce uncertainties on whale location, source IF, and clock drifts. Estimated whale location uncertainties are 30–160 m and clock drift uncertainties are 3–26 ms. The prior and posterior probability densities for environmental parameters are used to quantify transmission loss uncertainties corresponding to different levels of environmental knowledge, with applications to computing marine-mammal sound exposure levels.

1:35

4pAO3. Estimating source levels and depth distributions of calling bowhead whales using geoacoustic inversion. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Susanna Blackwell (Greeneridge Sci. Inc., Santa Barbara, CA), Katherine Kim (Greeneridge Sci. Inc., San Diego, CA), and A. Michael Macrander (Shell Exploration and Production Co., Anchorage, AK)

Every year during open-water season in 2008–2014, up to 40 passive acoustic recorders (DASARs) were deployed in the Beaufort Sea. Over the eight-year period, more than 3.1 million bowhead calls were localized. A vertical array was also deployed near DASARs in some years, enabling the matched-field geoacoustic inversion of bowhead whale calls to estimate the local acoustic propagation

environment. In turn, these environments were used to estimate the source level and calling depth distributions of nearly 50,000 calls between 2008 and 2014. The resulting source level distributions exhibited a stable mode near 160 dB re 1 uPa @ 1 m (rms), which proved robust to the year investigated, the whale call analysis method (manual or automated), and the type of acoustic propagation model (a simple empirically-derived "power-law" transmission loss model or a detailed waveguide propagation model). Depth distributions of calling whales were more sensitive to the environmental model used but always showed a mode at ~25 m depth. The evolution of the frequency distribution of calls over the years will also be discussed. This research highlights the importance of incorporating geoacoustic inversion and propagation modeling into marine mammal behavioral or density estimation studies. [Sponsored by Shell Exploration and Production Company.]

Contributed Papers

1:50

4pAO4. Applying double-difference methods for fine-scale acoustic tracking of controlled sources and sperm whales using a small aperture vertical array. Ludovic Tenorio-Hallé, Aaron M. Thode, Jit Sarkar (Scripps Inst. of Oceanogr., 1044 Loring St., San Diego, CA 92109, ludovictenorio@gmail.com), Chris Verlinden, Jeffrey D. Tippmann, William S. Hodgkiss, and William A. Kuperman (Scripps Inst. of Oceanogr., La Jolla, CA)

Ray propagation modeling can estimate a source's depth and range in a waveguide by exploiting multipath arrival information on a vertical array. However, environmental mismatch of the model, array tilt, and limited angular resolution of an array can yield highly scattered dive trajectories when ray tracing individual events. "Double-difference" methods have been used to localize earthquakes (Waldhauser and Ellsworth, 2000) and fin whales (Wilcock, 2012) by determining the location of multiple events relative to each other, rather than their absolute position. These same concepts can be reformulated into a "triple-difference" approach to track successive acoustic events on a single multi-hydrophone array. This method examines relative changes in the multipath arrival times and elevation angles over the course of a dive in order to establish a more robust track in terms of relative positions along the trajectory. Presented here are results of applying this new technique on both a towed source and sperm whales, using acoustic data recorded on a short aperture vertical array off the coast of Southern California in 4 km deep water. [Work supported by Office of Naval Research—Marine Mammals and Biology and Ocean Acoustics Program.]

2:05

4pAO5. Humpback whale-generated ambient noise levels provide insight into singers' spatial densities. Kerri D. Seger, Aaron M. Thode (Scripps Inst. of Oceanogr., 9331 Discovery Way, Apt. C, La Jolla, CA 92037, kseger@ucsd.edu), Jorge Urbán R., Pamela Martínez-Loustalot, M. E. Jiménez-López, and Diana López-Arzate (Universidad Autonoma de Baja California Sur, La Paz, Mexico)

Previous research has investigated whether diffuse ambient noise levels can be used to estimate relative baleen whale abundance in environments where their vocal activity dominates ambient noise levels. Presented here is an analytical model of ambient noise levels as generated by randomly distributed singing humpback whales. The model exploits earlier ones that derive ambient noise levels by assuming randomly distributed wind-driven breaking surface waves. Using a parametrized acoustic propagation environment and various assumptions about humpback whale singing behavior, the current model predicts ambient noise levels as a function of frequency and population size. It also predicts that the "sensitivity" of ambient noise levels to changes in population size should be relatively independent of an individual's singing behavior, but does depend strongly on the population's spatial density. Using visual survey and bottom-mounted recorder data from the 2013 and 2014 humpback whale breeding seasons off Los Cabos, Mexico, a generalized linear model estimated the sensitivity of song-generated ambient noise levels across several frequency bands, while compensating for diel cycles. The results indicate that whales generally space themselves evenly over a constantly expanding region, but may tolerate a slightly higher density as the number of participating singers increases.

2:20

4pAO6. Underwater acoustic vector sensor recording apparatus for soundscape measurements. Richard D. Lenhart and Jason D. Sagers (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, lenhart@arlut.utexas.edu)

An Autonomous Underwater Multi-Dimensional Acoustic Recorder (AUMDAR) for making directional soundscape measurements has been designed, constructed, and tested at The Applied Research Laboratories. The AUMDAR includes several three-dimensional transducers (acoustic vector sensor, gravitometer, accelerometer, and gyroscope), a multichannel recording unit (six channels, up to 96 kHz sampling rate and 24 bit resolution), and a battery power supply housed in a 90 m depth-rated pressure vessel. The apparatus can be deployed on the seafloor via an oceanographic tripod, suspended within the water column on a vertical mooring line, or tethered below a surface float. Laboratory-based acoustical performance characterization measurements including vector sensor on-axis receive sensitivity, directivity, and electrical noise floor are discussed. Field measurements recorded by the AUMDAR in Lake Travis, TX, and Haro Strait, WA, are also presented.

2:35–2:55 Break

2:55

4pAO7. Modelling the performance of fish tag monitoring stations on the Scotian Shelf. Danielle Moore and David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca)

The Ocean Tracking Network operates and maintains a continental shelf scale array of 256 bottom mounted fish-tag monitoring stations, spanning from the entrance of Halifax harbour to the Scotian Shelf break. These stations detect tagged keystone, commercially important, and endangered species as they migrate across this acoustic curtain, known as the Halifax line. The detection performance of each station is dependent on the local bathymetry, oceanography (sound speed profile variability), ambient noise level, and source depth distribution. At each station, local sound speed profiles from archived glider data were collected and sorted into representative groups. An Nx2D ray-trace model was used to calculate the transmission loss and relative strength of the noise field at fish tag frequencies (69 kHz) for each of the representative sound speed profiles at a number of stations across the array. The performance variability at each station and between stations is presented and compared to real detection data.

3:10

4pAO8. The effects of bottom sediments on the measured spectrum of seismic airgun pulses as a function of range. Bruce Martin, Jeff MacDonnell, and Loren Bailey (JASCO Appl. Sci., 32 Troop Ave., Ste. 202, Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com)

Sound from seismic airgun pulses is well known to be a measurable component of the marine soundscape even hundreds of kilometers from the source vessel. The peak frequency of the pulses is normally below 100 Hz, and hence the concern is often raised that the airguns may mask the communications of large baleen whales. Recently, there have been reports of significant energy from seismic airguns in the range of 1–10 kHz which could mask calls from a wider range of marine life. Measurements and modeling of a variety of deep water environments are compared to show that environments with very hard bottom types support the long-range propagation of frequencies above 500 Hz from seismic airgun arrays.

Invited Papers

3:25

4pAO9. Broadband active acoustics for synoptic studies of marine ecosystem. Andone C. Lavery and Timothy K. Stanton (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

Marine ecosystems host a wide variety of organisms spanning many trophic levels, ranging from the smallest planktonic organisms that comprise the base of the marine food web, to large apex predator fishes, marine mammals, such as whales, and sea birds. Active acoustical systems that span a wide range of frequencies are required to quantitatively characterize these organisms, their distribution and abundance. In this presentation, we discuss the development and use of broadband acoustic systems that are optimized for studies of fish and zooplankton through use of spectral classification methods. For swimbladder-bearing fish, a low-frequency (1–6 kHz) broadband system is described that classifies fish according to the resonance of its swimbladder. For zooplankton, a high frequency (25–600 kHz) broadband system is described that classifies zooplankton either according the resonance of their gas (gas-bearing zooplankton only) or according to the transition between the Rayleigh and geometric scattering region (non-gaseous zooplankton). Because of the resolving power of the broadband signals, the echoes also tend to be non-Rayleigh, which has additional classification information. Applications of the spectral and statistical broadband acoustics methods to ecosystem research are given. [Work supported by the Office of Naval Research and the Woods Hole Oceanographic Institution.]

3:40

4pAO10. Fully utilizing the acoustic record for biological monitoring and ecological applications. John K. Horne, Ross Hytnen (School of Aquatic and Fishery Sci., Univ. of Washington, 1122 NE Boat St., Seattle, WA 98105, jhorne@uw.edu), Shari Maxwell (Pacific Northwest National Lab., Sequim, WA), Kenneth Ham (Pacific Northwest National Lab., Batelle, WA), Adam Maxwell (Pacific Northwest National Lab., Sequim, WA), and Jeff Condiotty (Simrad Fisheries, Lynnwood, WA)

Increasing demands to monitor marine ecosystems amplifies the need for efficient and effective characterization of organism abundances and distributions. Monitoring, processing, and archiving acoustic data may also be required in near-real-time when documenting environmental change or reporting impacts when meeting regulatory requirements. The Nekton Interactive Monitoring System (NIMS) is designed to address four challenges of ocean observation and monitoring: animal tracking, distribution characterization, regulatory thresholds, and data volume reduction. A Kalman filter is used to identify and link candidate targets into tracks. Backscatter measures are characterized using a suite of metrics (Sv, inertia, dispersion, Aggregation Index, evenness, % occupied) to quantify vertical distributions of pelagic organisms. If tracks cross pre-set exclusion ranges or regulatory thresholds of metrics are exceeded, then notifications are sent to trigger operational modification or mitigation. Kongsberg M3 sonar raw data acquisition rates of 11 Mb per second are reduced to 11 Mb per hour data storage, a 4 order of magnitude data savings. NIMS middleware can be deployed autonomously with instrument packages, remotely to telemeter data, network connected for real time monitoring, or used to process archived data. Example applications include ocean observatories and marine renewable energy environmental monitoring.

3:55

4pAO11. Application of acoustic technologies to study the temporal and spatial distributions of the Pacific hake (*Merluccius productus*) in the California Current System. Dezhang Chu, Rebecca Thomas (NOAA Fisheries, NWFSC, 2725 Montlake Blvd. E., Seattle, WA 98112, dezhang.chu@noaa.gov), Julia Clemons (NOAA Fisheries, NWFSC, Newport, Oregon), Sandy Parker-Stetter, John Pohl (NOAA Fisheries, NWFSC, Seattle, WA), Julia Clemons (NOAA Fisheries, NWFSC, Newport, Oregon), and Stephane Gauthier (Fisheries and Oceans Canada, Inst. of Ocean Sci., Sidney, BC, Canada)

Advances in acoustics technologies offer a remote and non-invasive sensing means to conduct fisheries acoustic surveys. Over the past two decades, joint US and Canada acoustic and trawl surveys on Pacific hake (*Merluccius productus*), one of the most important commercial fisheries off the West Coasts of the United States and Canada, have been conducted at the intervals of one to three years within the California Current System (CCS). In this presentation, the temporal and spatial distributions of Pacific hake resulting from these surveys spanning a period of nearly two decades will be presented. Challenges in converting the measured acoustic quantities to biological quantities, such as abundance and biomass, will be addressed, including uncertainties associated with mixed species, environmental parameters, and properties in fish morphology and anatomy. Issues related to transitions from single-species to ecosystem-based acoustic surveys will also be discussed.

Contributed Papers

4:10

4pAO12. High-frequency broadband acoustic backscatter from phytoplankton. Dylan L. DeGrâce (Oceanogr., Dalhousie Univ., 6299 South St., Halifax, NS B3H4R2, Canada, dylan.degrace@dal.ca) and Tetjana Ross (Inst. of Ocean Sci., Sidney, BC, Canada)

Current methods in phytoplankton detection and monitoring are often limited by low temporal and spatial resolution. In principle, the use of a high-frequency broadband acoustic system would be advantageous when used in conjunction with current methods; providing improvements both temporally and spatially. With this motivation, a high-frequency broadband active acoustic system has been developed and used in four separate trials to measure the backscatter from four morphologically-distinct species of phytoplankton. The morphologies studied include (1) a siliceous shelled cylinder, (2) a chain-forming siliceous shell cylinder, (3) a fluid-like spheroid, and (4) a soft-shelled spheroid; and whose sizes range from 10 to 60 μm. Organism cultures were insonified at frequencies between 0.75 MHz and 6.9 MHz giving a *ka* study range of 0.03–1.73. Volume scattering strength as it varies with *ka* is presented for each species and compared to potential scattering modelsdrawn from the zooplankton scattering literature. Modifications to the models or model parameters are discussed. Additionally, volume scattering strengths at multiple phytoplankton concentrations are

4p THU. PM

presented and compared to both chlorophyll-*a* estimates obtained from fluorometers and densities found via flow cytometry. The potential for a phytoplankton species-detection and monitoring system is discussed and evaluated.

4:25

4pAO13. Measurements of the acoustic properties of the seagrass *Posidonia oceanica*. Jay R. Johnson, Gabriel R. Venegas, Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, johnson.jayrichard@utexas.edu), and Jean-Pierre Hermand (Université libre de Bruxelles, LISA - Environ. HydroAcoust. Lab, Brussels, Brussels Capital, Belgium)

A one-dimensional resonator technique was used to test the acoustic response of fresh leaves of *Posidonia oceanica* collected from both Crete, Greece and Sicily, Italy. The sound speed was inferred from resonance frequencies measured between 1 and 8 kHz. This sound speed is also compared with ultrasonic time-of-flight measurements between 1 and 4 MHz. Measurements of intact leaves as well as a homogenous tissue "soup" were made to investigate tissue properties and the role of leaf structure in sound propagation. This work expands on similar measurements, published previously by the authors, made on three other seagrass species *Thalassia testudinum, Halodule wrightii,* and *Syringodium filiforme*.

4:40

4pAO14. Bioacoustic absorption spectroscopy: An acoustical oceanography method for the study of marine ecosystems. Orest Diachok (Johns Hopkins Univ. APL, 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com)

Recent experiments have demonstrated the potential of the Bioacoustic Absorption Spectroscopy (BAS) method for measuring number densities vs. fish length and species when combined with concurrent trawling data, and studying vertical migrations and schooling behavior of fish at mesoscale dimensions.. Measurements of transmission loss between moving or stationary broadband source(s) and vertical array(s) of hydrophones separated by several (up to ~10) km permits inference of absorption coefficients versus frequency and depth, and association of absorption lines with the resonance frequencies of fish swim bladders. BAS is environmentally friendly, as it can be implemented with source levels < 170 dB, is sensitive to fish throughout the water column, including near boundaries, and is not affected by *avoidance*. The most significant results of BAS experiments will be reviewed, and possible approaches for use of the BAS method for routine scientific study and fisheries surveys of marine ecosystems will be considered.

4:55

4pAO15. Active acoustic monitoring in extreme turbulence around marine renewable energy devices. Shaun Fraser (School of Eng., The Univ. of Aberdeen, Fraser Noble Bldg., Aberdeen AB24 3UE, United Kingdom, s.fraser@abdn.ac.uk), Benjamin Williamson, Beth E. Scott (School of Biological Sci., The Univ. of Aberdeen, Aberdeen, United Kingdom), and Vladimir Nikora (School of Eng., The Univ. of Aberdeen, Aberdeen, United Kingdom)

The advance of tidal energy technologies has created new demands for active acoustic monitoring in highly dynamic marine environments. An innovative data collection approach using the FLOWBEC multi-instrument platform has been developed to acoustically observe turbulence and ecological interactions in the challenging environments around turbine installations in the UK. Standard processing approaches for echosounder data are unsuitable in these sites because of the extreme variability in acoustic conditions due to strong tidal flows and complex wind-wave interactions. Novel techniques for identifying ecological targets (fish, diving seabirds, and marine mammals) and characterising the physical conditions have been developed which are functional even during extreme turbulence. Reliable target identification is achieved using scale-sensitive filtering, morphological characterization, and multifrequency analysis of EK60 echosounder data. Combining results with synchronized multibeam data and other observations gives new oceanographic and ecological insights into these environments. This study contributes novel methodological and processing concepts for acoustic analysis in challenging sites of emerging industrial importance. The results provide vital observations on the behaviour of marine species with clear applications for the analysis of environmental impacts of marine renewable energy technologies.

5:10

4pAO16. An evaluation of the frequency response of hydrocarbon droplets. Scott Loranger and Thomas C. Weber (Univ. of New Hampshire, Jere A Chase Ocean Eng., 24 Colovos Rd., Durham, NH 03824, sloranger@ccom.unh.edu)

Development of instrumentation to detect and quantify submerged oil droplets would provide researchers and oil spill responders with crucial information about the fate and movement of oil in the environment. By detecting oil droplets in the watercolumn it should be possible to trace surface sheens to their source and to determine the location and extent of plumes of oil at depth. Methods of detecting oil currently exist, for example, mass spectrometers and fluorometers; however, they are limited to detecting oil that is submeter range from the instrument. Using broadband high frequency (30–300 kHz) acoustic echosounders, it is possible to not only detect oil droplets from a greater distance (10s of meters for individual droplets, depending on the background noise) but to quantify the physical properties of the oil, including the size of droplets. Droplet size is an important factor in determining the likely location of submerged plumes and surface sheens, the rate of biodegradation and rise rate of oil. Laboratory measurements of the broadband response along with the sound speed, density and droplet size of crude oil, diesel, gasoline, and kerosene have been made. The frequency response of the droplets have been compared to models for the target strength of fluid filled spheres to verify the models, and to empirically derive adjustments if necessary. The data are also used to empirically estimate a detection range limit for different densities of droplets determined.

5:25

4pAO17. Time-difference-of-arrival localization of bowhead whales using asynchronous recorders. Graham A. Warner, Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd. Ste. A405, Victoria, BC V8P 5C2, Canada, gwarner@uvic.ca), and David E. Hannay (JASCO Appl. Sci., Victoria, BC, Canada)

This paper estimates bowhead whale locations and uncertainties using linearized Bayesian inversion of the time-difference-of-arrival (TDOA) of whale calls recorded on omni-directional asynchronous recorders in the Chukchi Sea, Alaska. A Y-shaped cluster of seven autonomous ocean-bottom hydrophones, separated by 0.5–9.2 km, was deployed for several months over which time their clocks drifted out of synchronization. Hundreds of recorded whale calls are manually annotated with time-frequency bounds and associated between recorders. The TDOA between all hydrophone pairs are calculated from filtered waveform cross-correlations and depend on the whale locations, hydrophone locations, relative recorder clock drifts, and an effective waveguide sound speed. The inversion estimates all of these parameters and their uncertainties as well as data error statistics using prior information to constrain the otherwise underdetermined problem. Whale location uncertainties are estimated to be approximately 100 m which allows tracking whales that vocalize repeatedly over several minutes. Estimates of clock drift rates are obtained from inversions of TDOA data over several weeks. The inversion is computationally efficient and suitable for application to large datasets of manually- or automatically detected whale calls.

Session 4pBA

Biomedical Acoustics: Imaging

Kausik Sarkar, Chair
George Washington University, 801 22nd Street NW, Washington, DC 20052

Contributed Papers

1:30

4pBA1. Simulation and Implication of a two-dimensional phased array flexible ultrasound system for tissue characterization. Zhihua Gan (Biomedical Eng., Stony Brook Univ., Rm. 212A, BioEng. Bldg., Stony Brook, NY 11794, zhihua.gan@stonybrook.edu), Brian Guo (Dynamic Res. Instruments LLC, Stony Brook, New York), Jiqi Cheng, and Yi-Xian Qin (Biomedical Eng., Stony Brook Univ., Stony Brook, NY)

It has been demonstrated that phased-array ultrasound can be used for configuring spatial focal zone and perform noninvasive characterization of trabecular bone quality. Such a sensor configuration can also be integrated into existing ultrasound triggering, data analysis, mux, and imaging system in a flexible ultrasound platform. The primary goal of this study is focused on the unique dual 2-D array transducer kit design with 1MHz center frequency and simulation to a flexible ultrasound system for trabecular bone assessment. Transmission (Tx) and receiving (Rx) transducers are designed as two identical 2-D array sensors, with elements arrayed in 27*27 as a square. Element width is 1.05 mm and kerf is 0.2 mm. Tx elements are divided into sub-blocks to excite ultrasound signal in sequence to decrease the system complexity while maintaining beam pattern properties by the signal processing procedure at Rx side. Appropriate delay modules are inserted into the process of exciting signal. The working procedure is simulated by FIELD II program. The gray-scale simulation of acoustic resolution and side lobes are analyzed. The theoretical lateral resolution (LR) is 2.436 mm, with 2.8 mm in simulation. Hann window based apodization process shows the reduced side lobes. The results suggested that the reduced array size (27*27) simplifies the design and manufacturing, but maintains the image/signal resolution.

1:45

4pBA2. A stable numerical method for wide-angle parabolic models of focused ultrasound. Joshua Soneson (Div. of Appl. Mech., US Food and Drug Administration, 10903 New Hampshire Ave., Silver Spring, MD 20993, joshua.soneson@fda.hhs.gov)

Wide-angle parabolic equations offer an attractive balance of utility, speed, and ease of implementation in the computation of sound beams in the frequency domain. These models are characterized by representing the pseudodifferential operator of the one-way Helmholtz equation with a Padé approximation, which is a quotient of polynomial operators, to which Crank-Nicolson-type numerical methods are easily applied. However, standard methods for determining the coefficients of these polynomials yield numerical schemes whose solutions exhibit oscillatory artifacts when source discontinuities are present, as is often the case for ultrasound transducers. In this work, two of three polynomial coefficients in a Padé approximation are used to obtain second order accuracy while the third coefficient is used to optimize stability. This results in a numerical method which effectively damps the spurious oscillations but retains the wide-angle capability, bringing to medical ultrasound a simple way to rapidly compute shallowly focused or steered beams.

2:00

4pBA3. Development of a nonlinear model for the pressure dependent attenuation and sound speed in a bubbly liquid and its experimental validation. Amin Jafari Sojahrood (Dept. of Phys., Ryerson Univ., 350 Victoria St., Toronto, ON M5B2K3, Canada, amin.jafarisojahrood@ryerson.ca), Qian Li, Mark Burgess (Mech. Eng., Boston Univ., Boston, MA), Raffi Karshafian (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada), Tyrone Porter (Mech. Enginnering, Boston Univ., Boston, MA), and Michael C. Kolios (Dept. of Phys., Ryerson Univ., Toronto, ON, Canada)

Presence of the MBs in the sound field increases the attenuation of the medium and changes the sound speed. A detailed knowledge about the attenuation of the medium is critical for controlling and optimizing the behavior of the MBs in applications. However, existing models of ultrasound attenuation in bubbly mediums are based on linear approximations (low amplitude MB oscillations) and thus are not valid in many regimes used in applications. A model to calculate the nonlinear attenuation and sound speed is developed by deriving the complex and real part of the wave number from the Calfish model. The predictions of the model were validated by measuring the attenuation and sound speed of dilute monodisperse MB solutions (5000 microbubbles/ml) with median diameter of 5.2 and 9.8 μm using acoustic pressure range of 10–130 kPa. The attenuation of the medium was calculated by numerically solving the radial oscillations of the MB and incorporating it in the attenuation model. Predictions of the model were in good agreement with the experimental results. As the acoustic pressure increased, the attenuation and maximum sound speed of the medium increased from 5 dB/cm to 12 dB/cm and 1500 to ~1530 m/s, respectively.

2:15

4pBA4. A stethoscope for the knee: Investigating joint acoustical emissions as novel biomarkers for wearable joint health assessment. Omer T. Inan, Sinan Hersek, Caitlin N. Teague, Hakan Toreyin, Hyeon K. Jeong (Elec. and Comput. Eng., Georgia Inst. of Technol., Technol. Square Res. Bldg., 85 Fifth St NW, Ste. 412, Atlanta, GA 30308, inan@gatech.edu), Michael L. Jones, Melinda L. Millard-Stafford, Geza F. Kogler, and Michael N. Sawka (Appl. Physiol., Georgia Inst. of Technol., Atlanta, GA)

Each year, millions of Americans endure knee injuries, ranging from simple sprains to ligament tears requiring surgical intervention. Our team is investigating wearable rehabilitation assessment technologies for patients recovering from knee injuries based on the measurement and analysis of the acoustical emissions from the knees. Using miniature electret microphones combined with piezoelectric sensors placed on the surface of the skin at the knee, we measure the sounds from the joint as subjects perform basic flexion/extension exercises and standardized sit-to-stand protocols. We then analyze the consistency of the knee acoustical emissions in the context of the activity, and the angle of the joint, to quantify the health of the joint. We have found, in early pilot studies, promising results differentiating the healthy versus injured knee, and longitudinal changes progressing from acute injury and recovery following rehabilitation. We have also determined

that, in healthy subjects, the pattern of acoustic emissions is consistent within several repetitions of a movement, and for multiple recordings throughout the day ($r > 0.88$). Knee acoustic emissions combined with angle measurements provide promising in-depth information regarding joint health and exciting new opportunities for personalized rehabilitation protocols following injury.

2:30

4pBA5. Photoacoustic imaging of muscle oxygenation during exercise. Clayton A. Baker, Nashaat Rasheed, Parag V. Chitnis, and Siddhartha Sikdar (BioEng., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, cbaker6@masonlive.gmu.edu)

Monitoring muscle hemodynamics and oxygenation is important for studying muscle function and fatigue. Current state-of-the-art for noninvasive oximetry is near-infrared spectroscopy (NIRS), which provides relative oxygen saturation (SO_2) with good sensitivity but has poor spatial resolution and sensing depth, and does not provide anatomical context. In this study, we demonstrated the utility of a dual-modality imaging system that could generate co-registered ultrasound (US) and photoacoustic (PA) images for real-time, functional imaging of human muscle. The system consisted of a wavelength-tunable pulsed laser (Opotek) integrated with a research ultrasound system (Verasonics). US images provided anatomical context and the PA images acquired at 690 nm and 830 nm were processed to estimate SO_2 during a sustained isometric contraction and return to rest using a protocol approved by our Institutional Review Board. PA-based SO_2 was compared to measurements acquired from a commercially available NIRS oximeter under the same conditions. Preliminary results showed good qualitative agreement between SO_2 dynamics observed via PA-based approach and NIRS, with contraction coinciding with an exponential decay in SO_2, and relaxation with a return to original levels. This *in vivo* study demonstrated the feasibility of PA imaging for measuring temporally and spatially resolved muscle oxygenation during functional tasks.

2:45

4pBA6. Clinical results of ultrasound bladder vibrometry for assessment of bladder compliance. Mahdi Bayat, Mathew Cheong, Max Denis, Viksit Kumar (Physiol. and Biomedical Eng., Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, bayat.mahdi@mayo.edu), Mohammad Mehrmohammadi (Biomedical Eng., Wayne State Univ., Detroit, MI), Adriana Gregory, Ivan Nenadic (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN), Douglas Husmann, Lance Mynderse (Urology, Mayo Clinic, Rochester, MN), Azra Alizad, and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Healthy human bladder is a viscoelastic shell capable of expanding to allow storage of urine at low pressures. The ability to maintain low pressures as the urine volume increases is called bladder compliance. Compliance is a key in proper functioning of urinary system. Spinal cord injuries and other neurogenic diseases can cause deterioration in bladder expansion capability (e.g., through excessive growth of fibrotic tissue), which in turn can increase the rigidity of the wall and reduce bladder compliance. Urodynamic study (UDS) is the clinical standard for evaluation of the bladder compliance through observation of bladder pressure-volume behavior. However, this method is costly and invasive. Ultrasound bladder vibrometry (UBV) has been introduced as noninvasive technique for evaluation of bladder viscoelasticity through excitation of the bladder using ultrasound radiation force and tracking the resulting Lamb waves. Here, we present the results of UBV from 44 patients undergoing concurrent UDS and UBV studies. Our results show that elasticity parameters obtained from the UBV closely correlate with the UDS pressure-volume data with a median Pearson's correlation value of more than 0.8. These results prove the potential utility of UBV as an alternative noninvasive method for bladder compliance assessment. [Work supported by NIH grant DK99231.]

3:00

4pBA7. Phantom study on the detectability of micro-tumors in breast tissue using high-frequency ultrasound. Nicole Cowan (BioTechnol., Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, ncowan18@gmail.com), Zachary A. Coffman (Biology, Utah Valley Univ., Orem, UT), Robyn K. Omer (Botany, Utah Valley Univ., Orem, UT), Benjamin F. Finch (Biology, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The ability to detect malignant tissue in surgical margins during breast cancer surgery would reduce the risk of local recurrence and the need for subsequent surgeries to remove residual cancer. A surgical study conducted by Utah Valley University with the Huntsman Cancer Institute showed that high-frequency ultrasound (20–80 MHz), and the parameters peak density (number of spectral peaks and valleys from 20 to 80 MHz) and attenuation, are sensitive to breast tissue pathology. Pathology results from this surgical study showed that many margin specimens contained micro-tumors measuring 1 mm in diameter or smaller. The present study's objective was to determine the sensitivity of high-frequency ultrasound to these micro-tumors using phantoms. Phantoms were created from distilled water, agarose powder, 10X TBE stock solution, and 390–925μm diameter polyethylene microspheres to simulate breast tumors. Microspheres were embedded in phantoms singularly and in clusters of 3–12 microspheres. Pitch-catch measurements were acquired using large (6.35 mm diameter) and small (1.5 mm diameter) 50-MHz transducers, a high-frequency ultrasound system, and glycerol as the coupling agent. Both large and small transducers were sensitive to single microspheres and microsphere clusters across all microsphere diameters. The phantom results confirm the sensitivity of high-frequency ultrasound to breast cancer micro-tumors and validate the surgical study.

3:15

4pBA8. High-frequency ultrasonic measurements of ischemia and revascularization in mice. Michaelle A. Cadet (Biology, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, michaelle.alexandra@gmail.com), Andrea N. Quiroz (Nursing, Univ. of Miami, Miami, FL), Andrew Chappell (Dentistry, Univ. of Louisville, Louisville, KY), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The ability to rapidly determine the degree of vascularization in small animals *in vivo* would provide a useful characterization tool for regenerative medicine. This study's objective was to determine if ultrasonic material property measurements in the 10–100 MHz range could be used as a vascularization assay for small animals. The study was performed at the Ludwig Boltzmann Institute for Experimental and Clinical Traumatology (Vienna, Austria), where the femoral artery in one hind limb of each of sixteen mice was ligated and tested over a time period of eight days. Eight of the ligated limbs were treated with vascular endothelial growth factor (VEGF). The remaining eight ligated limbs were allowed to grow ischemic. The unligated limbs were controls. Wave speed, attenuation, waveform coda amplitude, and spectral measurements were acquired using a high-frequency ultrasound system. Ischemic limbs displayed a steady decrease in wave velocity over the test period as compared to VEGF-treated limbs. Coda amplitude increased for ischemic limbs, but decreased and then returned to normal in VEGF-treated limbs. No trends were observed for either attenuation or spectral peak density. The results indicate that high-frequency ultrasonic measurements may provide an added dimension to small animal imaging methods for detecting revascularization.

3:30–3:45 Break

3:45

4pBA9. Effect of mammographic breast density on high-frequency ultrasonic parameters used to evaluate surgical margins. Zachary A. Coffman (Biology, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, zachary.a.coffman@gmail.com), Nicole Cowan (BioTechnol., Utah Valley Univ., Orem, UT), Robyn K. Omer (Botany, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Breast density, the proportion of connective tissue versus fat tissue in the breast, is typically determined using mammography. Women with higher breast density are four to five times more likely to develop breast cancer than women with lower densities. A surgical study performed by Utah Valley University with the Huntsman Cancer Institute showed that high-frequency ultrasound (20–80 MHz), and the parameters peak density (number of spectral peaks and valleys from 20 to 80 MHz) and attenuation, are sensitive to breast tissue pathology. This study also showed that breast density had no effect on peak density across the entire breast density range while attenuation increased 2X from entirely fat to extremely dense. The present study's objective was to determine the effect of breast density on these parameters using histology mimicking phantoms. Phantoms were created from distilled water, agarose powder, 10X TBE stock solution, and polyethylene microspheres and fibers to simulate breast tissue histology. Phantoms were produced with microspheres only, fibers only, and a combination of microspheres and fibers. Pitch-catch measurements were acquired using a high-frequency ultrasound system and glycerol as the coupling agent. The phantom results validate the surgical study, showing attenuation increasing with inclusion concentration but no significant change in peak density.

4:00

4pBA10. Interpreting attenuation at different excitation amplitudes to estimate strain-dependent interfacial rheological properties of monodisperse contrast microbubbles. Kausik Sarkar, Lang Xia (George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu), and Tyrone Porter (Boston Univ., Boston, MA)

Broadband attenuation of ultrasound measured at different excitation pressures being different raises a serious theoretical concern, because the underlying assumption of linear and independent propagation of different frequency components nominally requires attenuation to be independent of excitation. This issue is investigated by examining ultrasound attenuation through a monodisperse lipid coated microbubble suspension measured at four different acoustic excitation amplitudes. We use the attenuation data to determine interfacial rheological properties (surface tension, surface dilatational elasticity and surface dilatational viscosity) of the encapsulation according to three different models. Although different models result in similar rheological properties, attenuation measured at different excitation levels (4–110 kPa) leads to different values for them; the dilatation elasticity (0.56 N/m to 0.18 N/m) and viscosity (2.4×10^{-8} Ns/m to 1.52×10^{-8} Ns/m) both decrease with increasing pressure. Numerically simulating the scattered response, nonlinear energy transfer between frequencies are shown to be negligible, thereby demonstrating the linearity in propagation and validating the attenuation analysis. There is a second concern to the characterization arising from shell properties being dependent on excitation amplitude which is not a proper constitutive variable. It is resolved by arriving at a strain-dependent rheology for the encapsulation.

4:15

4pBA11. Quantitative microultrasound characterization of gastrointestinal tissue for ultrasound capsule endoscopy. Holly Lay (School of Eng., Univ. of Glasgow, Glasgow G12 8QQ, United Kingdom), Benjamin F. Cox (Ninewells Hospital and Med. School, Dundee, United Kingdom), Christine E. Demore (School of Eng., Univ. of Glasgow, Glasgow, United Kingdom), Gabriel C. Spalding (Illinois Wesleyan Univ., Bloomington, IL), and Sandy Cochran (School of Eng., Univ. of Glasgow, Glasgow, United Kingdom, sandy.cochran@glasgow.ac.uk)

Gastrointestinal (GI) disease progression is often characterized by cellular and architectural changes within the mucosal and sub-mucosal layers. One relevant disorder, Barrett's esophagus, is of particular interest as a recognized predictor of esophageal cancer. To enhance the clinical ability to detect cellular changes deeper in tissue and earlier, we are exploring quantitative microultrasound) techniques in healthy *ex vivo* porcine GI tissue for implementation in ultrasound capsule endoscopy. A single-element, piezocomposite mUS transducer operating at fc = 47.7 MHz was used to obtain pulse-echo images of *ex vivo* porcine gastroesophageal samples, which were bisected and mechanically scanned along the long GI axis. Selected samples were mechanically separated to isolate the upper mucosal layers from the underlying muscle layers and placed on a known agar substrate to calculate the attenuation coefficient of the tissue. All other sample data were digitally segmented. Reflectivity data from the top 50, 100, and 250 μm of tissue were analysed to assess the effect of variable surface density on the calculated acoustic impedance. The entire thickness of the segmented tissue was then used to calculate position-dependent backscatter coefficient (BSC) along with intra- and inter-sample variability for use as a baseline from which to make further quantitative advances.

4:30

4pBA12. Prediction of multivalued waveforms in media with power-law attenuation. John M. Cormack and Mark F. Hamilton (Appl. Res. Lab., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

The lossless Burgers equation predicts a multivalued waveform beyond a certain propagation distance. Inclusion of thermoviscous attenuation, which increases as frequency squared, prevents the occurrence of multivalued waveforms. The same is true for any attenuation law that is proportional to frequency raised to an exponent greater than unity. For exponents less than unity the situation is less clear. For example, when attenuation is constant with frequency (exponent equal zero) there is a critical value of the attenuation coefficient below which a multivalued waveform is predicted and above which it is not. To investigate the prediction of multivalued waveforms for power-law attenuation with exponents between zero and unity, a Burgers equation with the loss term expressed as a fractional derivative is used [Prieur and Holm, J. Acoust. Soc. Am. **130**, 1125 (2011)]. Transformation of the equation into intrinsic coordinates following Hammerton and Crighton [J. Fluid Mech. **252**, 585 (1993)] permits numerical solutions to be obtained that are used to determine the parameter space in which initially sinusoidal plane waves are predicted to evolve into multivalued waveforms for power-law attenuation with exponents less than unity. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

4:45

4pBA13. Evaluating lymph node status with high-frequency ultrasound during breast conservation surgery. Amy A. LaFond (Biology, Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, fairbrother.aa@gmail.com), Caitlin Carter (BioTechnol., Utah Valley Univ., Orem, UT), Dolly A. Sanjinez (Biology, Utah Valley Univ., Orem, UT), Robyn K. Omer (Botany, Utah Valley Univ., Orem, UT), Leigh A. Neumayer (Surgery, Univ. of Arizona, Tucson, AZ), Rachel E. Factor (Pathol., Univ. of Utah, Salt Lake City, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

High-frequency (20–80 MHz) ultrasound was used to evaluate 78 lymph nodes from 39 patients to identify metastatic breast cancer during breast conservation surgery. Point measurements were collected from resected lymph nodes in both through-transmission and pulse-echo modes using 50-MHz, 6.35-mm diameter, single-element transducers. Attenuation and peak density (the number of peaks and valleys in a specified frequency band) were calculated from the ultrasonic waveforms and power spectra, respectively. The two parameters were additionally combined to perform a multivariate analysis. Fisher's exact test was used to determine the accuracy, sensitivity, and specificity of each parameter and the multivariate analysis for detecting malignant lymph nodes. The multivariate analysis showed the greatest statistical measures, with an 83.3% accuracy, 87.5% sensitivity, 82.9% specificity, and a p-value of 0.000078 (high statistical significance). The results demonstrate that high-frequency ultrasound provides very good sensitivity and specificity for malignant lymph nodes in the breast, and that high-frequency ultrasound is a viable, prospective method to be used as a rapid, intraoperative, and potentially*in vivo*diagnostic tool by surgeons for a wide range of soft tissue cancers.

5:00

4pBA14. Detecting changes in the cytoskeletal structure of malignant pancreatic cells using high-frequency ultrasound. Caitlin Carter (Bio-Technol., Utah Valley Univ., 800 W. University Parkway, MS 179, Orem, UT 84058-5999, caitlin.carter03@gmail.com), Ashley Behan, Dolly A. Sanjinez, Amy A. LaFond (Biology, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Traditional classification of breast cancer subtype is determined based on protein expression and genetic profile. Determining the molecular subtype of breast cancer can make prognosis more accurate and help to personalize treatment. More aggressive subtypes of breast cancer, such as basal-like and Her2+, have mutations that alter the protein regulation of the cell cytoskeleton. These cytoskeleton modulations can enable the cell to become more mobile and metastasize to other parts of the body. High-frequency ultrasound (10–100 MHz) has previously been studied for the real-time diagnosis of malignant tissue in breast conservation surgery. High-frequency ultrasound also has the potential of determining the molecular subtype of breast cancer. The objective of this work was to determine if chemically induced changes in the cytoskeleton of cancer cell lines are able to be detected using high-frequency ultrasound. Cell cultures of a human pancreatic carcinoma cell line (panc-1) were grown in monolayers and then treated with sphingosylphosphorylcholine (SPC), a bioactive lipid that rearranges the keratin components of the cytoskeleton. Pulse-echo measurements of the cultures were taken over a period of one hour. The results of this work demonstrate that high-frequency ultrasonic spectra are sensitive to the cytoskeleton changes induced by SPC.

THURSDAY AFTERNOON, 26 MAY 2016 SOLITUDE, 1:30 P.M. TO 6:00 P.M.

Session 4pMU

Musical Acoustics: Session in Honor of William J. Strong

Thomas Rossing, Chair
Stanford University, 26464 Taaffe Rd., Los Altos Hills, CA 94022

Chair's Introduction—1:30

Invited Papers

1:35

4pMU1. The impact of William J. Strong on the acoustics program at Brigham Young University. Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., N181 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

The early years of acoustics at Brigham Young University (BYU) include names such as Eyring, Fletcher, and Knudsen. However, for nearly 35 years, Bill Strong was the face of acoustics at BYU. As a faculty member, he contributed significantly to the maintenance and strengthening of acoustics at BYU, influencing many lives along the way. One of those lives influenced was the author. Beyond his positive influence in shaping many lives, Bill also contributed significantly to acoustics research throughout his career. Much of his work centered on the acoustics of speech and music. His text, *Music, Speech, Audio*, has been used for decades to introduce many students to the field of acoustics and has been the text for a popular course on the subject at BYU. This presentation will overview some of Bill's many contributions, both in acoustics and otherwise, with an emphasis on some of the work carried out by the author and Bill on the acoustics of the clarinet. That work includes a study of the interactions associated with the combined player-clarinet system, as well as an investigation of the directivity of the sound radiated from the clarinet.

2:00

4pMU2. How well can a human mimic the sound of a trumpet? Ingo R. Titze (National Ctr. for Voice and Speech, Univ. of Utah, 156 South Main St., Ste. 320, Salt Lake City, UT 84101-3306, ingo.titze@utah.edu)

The length of an uncoiled trumpet horn is more than 2 m, whereas the length of a human supraglottal airway is about 17 cm. The vibrating lips and the vibrating vocal folds can produce similar pitch ranges, but the resonators have vastly different natural frequencies due to the more than 10:1 ratio in airway length. Humans rarely attempt to tune more than one resonance to a harmonic because of the disparity between harmonic spacing and airway resonances spacing, but the brass-like timbre of a human "call" or "belt" can mimic that of a trumpet. A one-dimensional Navier-Stokes solution of non-steady compressible flow in soft-walled airways is used to show the similarities and differences in time-domain wave propagation and resulting flow and pressure frequency spectra along the airways. One striking structural similarity between the human instrument and the brass instrument is the shape of the airway directly above the sound source, known as the mouthpiece for brass and the epilarynx tube for the human airway. It plays a major role in shaping the source spectrum.

2:25

4pMU3. William J. Strong's musical instrument models. James W. Beauchamp (School of Music and Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824, jwbeauch@illinois.edu)

After studying acoustics with Harvey Fletcher at Brigham Young University (BYU) in the 1950s, Bill joined the computer music sound analysis/synthesis group led by Melville Clark, Jr., at the Massachusetts Institute of Technology in 1959. He presented his first musical acoustics paper, on clarinet tone synthesis, at the fall, 1964 ASA meeting in Austin. This was followed by a paper on oboe synthesis at the spring, 1965 meeting in Washington. These papers were elaborated on in two wind instrument synthesis papers he and Clark published in JASA in 1967. The pioneering method was called *spectral envelope synthesis*, although temporal envelope functions were also incorporated in the model. Parameters for this complex synthesis method were obtained from time-variant spectrum analyses done by David Luce (Ph.D. thesis, MIT, 1963). Later back at BYU, in the late 1970s and 1980s, Bill and his students began analyzing acoustic structures of wind instruments, culminating with Michael Thompson (M.S. thesis, BYU, 2000) and Bill publishing results on measurement and simulation of nonlinear propagation in a trombone in JASA in 2001. This clearly demonstrated that nonlinear propagation of waveforms in a pipe can produce very significant effects on the "brassiness" of a brass instrument's output sound.

2:50

4pMU4. Acoustical factors affecting the playability of brass wind instruments with side holes. D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk)

Almost all modern instruments of the brass wind family radiate sound through a single terminating bell. From the Middle Ages until the second half of the nineteenth century, lip-excited wind instruments with side holes were also in frequent use in both secular and sacred musical ensembles. The acoustical properties of labrasones with side holes, such as the cornetto, the serpent, the keyed bugle, and the ophicleide, resemble in may ways those of conventional brass instruments such as the trombone, but also have features which recall woodwind instruments like the oboe and saxophone. This paper presents some recent studies on the playability of labrasones with side holes, acknowledging the outstanding contributions of William J. Strong and his collaborators to the experimental study and modelling of both woodwind and brass instruments.

3:15

4pMU5. Acoustics research at Brigham Young University during the late 1960s. George Plitnik (Phys., Frostburg State Univ., 120 Compton Hall, Frostburg, MD 21532, gplitnik@frostburg.edu)

As Dr. Strong's first Ph.D. student in musical acoustics (I actually arrived one semester before he did), he took on the task of reorganizing the musical acoustics program as well as adding a research division in speech. The fact that these programs were, and continue to be, successful is a tribute to Dr. Strong's dedication and hard work.

3:40–3:55 Break

3:55

4pMU6. Conical reed instruments: A hierarchy of parameters. Jean Kergomard, Philippe Guillemain, Fabrice Silva, and Christophe Vergez (CNRS-LMA, 31 Chemin Joseph Aiguier, Marseille 13402, France, kergomard@lma.cnrs-mrs.fr)

Some aspects of sound production by wind musical instruments are rather well understood. A minimum model for reed cylindrical instruments was proposed in 1983 by Mc Intyre *et al*. When the reed dynamics and resonator losses are ignored, the mouthpiece pressure is a square signal. Three primary parameters are necessary: (i) the mouth pressure; (ii) the "valve" parameter (based on the reed opening and its stiffness); (iii) the length of the cylinder. The model gives a simplified shape of the waveform; the playing frequency and the amplitude are rather well predicted. For further spectrum details, it is necessary to add losses, therefore a secondary parameter, the radius. Furthermore other parameters influence the spectrum: reed dynamics, toneholes, vocal tract.... What happens for conical reed instruments? Considering the waveform of the mouthpiece pressure, a fourth primary parameter is the length of the missing part of the truncated cone. Similarly to cylindrical instruments, a secondary parameter is related to the losses depending either on the input or the output one, or on the apex angle: one of these three parameters allows the determination of the two others. A discussion is proposed concerning the different kinds of conical instruments.

4:20

4pMU7. High-resolution directivities of musical instruments and the human voice: Recent research collaborations of Bill Strong. Timothy W. Leishman, K. J. Bodon, and Jennifer K. Whiting (Phys. and Astronomy, Brigham Young Univ., N247 ESC, Provo, UT 84602, twleishman@byu.edu)

Over the past few years, Bill Strong has worked with other researchers at Brigham Young University to develop a feasible method to assess high-resolution directivities of played musical instruments and the human voice. The approach has produced numerous directivity balloons with 5-degree angular resolution and variable bandwidths, meeting the requirements of the AES56-2008 standard for loudspeaker measurements. This presentation will give an overview of the method, its validation, and some interesting results for several musical instruments and talkers. Data from the measurements are intended to enhance theoretical and practical efforts of professionals seeking to better understand and work with these live sources of sound.

4:45

4pMU8. Acoustics in a physics program in the 1990s. Daniel Ludwigsen (Kettering Univ., 1700 University Ave., Flint, MI 49504, dludwigs@kettering.edu)

To a recent graduate from a liberal arts college with majors in physics and music, Dr. Bill Strong presented a unique opportunity to work on a graduate degree in physics, with research in musical acoustics, using a computational approach. The historical tradition of acoustics within the discipline of physics was important to that young graduate student then, in the 1990s, and continues to be important to this member of academia twenty years later. The work of Bill Strong in shaping and sustaining acoustics at Brigham Young University was at a critical juncture. This presentation of the contemporary setting and my mentor's approach will offer a grad student's perspective, through an appreciative filter of years of experience as a professor.

5:10

4pMU9. From musical acoustics to noise control. David C. Copley (Caterpillar, PO Box 1875, Peoria, IL 61656, Copley_David_C@cat.com)

This presentation is a personal reflection of the influence Bill Strong had on the author's career in noise control. From first encounters with the decibel to experimental and numerical research in trombone acoustics, from simple textbook quizzes to demanding 20-page single-problem essays, the author shows the influence Bill Strong had on his education in acoustics which became the bedrock for a career in machinery noise control. The presentation will cite specific example of concepts, analyses and techniques found in musical acoustics—first encountered by the author in his studies with Bill Strong—and similarities to industrial noise control situations. The examples will demonstrate how an education in musical acoustic translates to practical applications in noise control engineering.

5:35

4pMU10. The effect of inharmonicity in piano tones. Brian E. Anderson and William J. Strong (Dept. of Phys. and Astronomy, Brigham Young Univ., BYU, N283 ESC, Provo, UT 84602, bea@byu.edu)

This presentation will review research conducted to determine the effect of inharmonic partials on the pitch of piano tones [JASA, **117**, 3268–3272 (2005)]. Synthetic piano tones were created based on recordings of an upright piano. One set of tones were made to have inharmonic partials that matched the partial frequencies of the piano tone recording, whereas another set of tones consisted of harmonic partials. Listeners compared a synthetic, inharmonic piano tone to a set of synthetic, harmonic piano tone with varying fundamental frequencies. They were asked to select the harmonic tone whose pitch best matched that of the inharmonic tone. The pitch of the inharmonic tones was perceived to be sharp relative to the harmonic tones (sharp in terms of the comparison of their respective fundamental frequencies).

Session 4pNS

Noise, ASA Committee on Standards, Animal Bioacoustics, Engineering Acoustics, and Physical Acoustics: Wind Turbine Noise II

Nancy S. Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton Street, Boston, MA 02118

Kenneth Kaliski, Cochair
RSG Inc, 55 Railroad Row, White River Junction, VT 05001

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Contributed Papers

1:30

4pNS1. Monitoring the acoustic effects of pile driving for the first offshore wind farm in the United States. Arthur E. Newhall, Ying T. Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 210 Bigelow Lab, MS #11, Woods Hole, MA 02543, anewhall@whoi.edu), James F. Miller, Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Kathy Vigness-Raposa, Adam Frankel, Jennifer Giard (Marine Acoust., Inc., Middletown, RI), Dennis R. Gallien, Jamey Elliot (HDR, Inc., Athens, AL), and Tim Mason (Subacoustech Environ. Ltd., Bishops Waltham, Hampshire, United Kingdom)

The Block Island Wind Farm, the first offshore wind farm in the United States, consists of five turbines in water depths of approximately 30 m. The turbines have a jacket type substructure with piles driven to the bottom to pin the structure to the seabed. A number of acoustic sensors were deployed to monitor the acoustic properties of the pile driving activity. The acoustic sensor systems consisted of an eight element towed hydrophone array, two fixed moorings with four hydrophones each, and a fixed sensor package for measuring particle velocity. The towed array was towed from 1 to 8 km from the pile driving location. The fixed moorings were deployed at 7.5 and 15 km from the pile location. The particle velocity sensor package was deployed at 500 m from the pile driving location. This sensor package consisted of a three-axis geophone on the seabed and a tetrahedral array of four low sensitivity hydrophones at 1 m from the bottom. Data collected on these sensor systems will be presented. Acoustic levels and particle velocity observations will be compared and their implications in the context of effects on marine life will be discussed. [Work supported by Bureau of Ocean Energy Management (BOEM).]

1:45

4pNS2. Guidelines for developing regulations for acoustic impact, based on the stage of operation of wind farms in Chile. Elias N. Montoya and Ismael P. Gomez (Universidad Tecnológica de Chile INACAP, Brown Norte 290, Ñuñoa, Santiago 7750000, Chile, enmontoyag@gmail.com)

Wind farms manifest different sonic characteristics than other sources, since the noise emitted depends on the wind which turns the blades and varies constantly, being able to tonal components, amplitude modulated, and low frequency noise. For these reasons, the international community has developed specific regulations to assess the acoustic impact associated. Nevertheless, the present Chilean noise regulations are not adjusted properly to the situation described above. Considering the exponential increase experienced by this energy source in recent years in Chile, the development of regulations for such features in the stage operation is suggested so that the noise impact can be properly addressed and quantified. Guidelines for the development of an eventual specific Chilean regulation will be proposed by analyzing and comparing international standards. From this work, we will determine the maximum noise levels, methodologies and other suggestions for the proposed target issues.

2:00

4pNS3. Measurements of infrasound blade pass frequencies inside multiple homes using narrowband analysis. Andy Metelka (Sound and Vib. Solutions Canada, Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Previous measurements in homes near wind turbines indicate higher pressure levels below 10Hz than audible pressure levels measured at the same time and location (Dooley and Metelka, ASA **20**, 2013). Long term measurements of Infrasound pressures appear inside multiple homes as wind speed and wind direction vary. Measuring Narrowband signatures identify fingerprints of rotational components of wind turbines. The measurements directly correlate how blade pass frequencies appear and disappear everywhere as wind changes. Data from four homes were measured and broadband infrasound levels from wind are compared to tonal infrasound blade pass frequencies In both cases, broadband infrasound and blade-to-tower pressures increase with wind and advanced signal processing techniques were used tracking signatures buried in noise.

2:15–4:15 Panel Discussion

Session 4pPAa

Physical Acoustics: General Topics in Physical Acoustics I

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

Contributed Papers

1:00

4pPAa1. Origin of negative density. Sam Hyeon Lee (Phys., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea), samlee@yonsei.ac.kr) and Oliver B. Wright (Div. of Appl. Phys., Hokkaido Univ., Sapporo, Japan)

We present a new physical interpretation for the effective densities. We introduce the idea of hidden force: the effective density is negative when the hidden force is larger than, and operates in antiphase to the applied force. We demonstrate this picture for some established acoustic metamaterials with elements based on membranes, springs, and masses. The hidden force for membrane-based acoustic metamaterials, for instance, is the force from the membrane tension. We also explain the analogous concepts for pure mass-and-spring systems, in which case, the hidden force can arise from masses and springs fixed inside other masses. This new picture provides a powerful tool for conceptual understanding and design of new acoustic metamaterials, and avoids common pitfalls involved in determining the effective parameters of such materials.

1:15

4pPAa2. Origin of negative modulus. Sam Hyeon Lee (Phys., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, samlee@yonsei.ac.kr) and Oliver B. Wright (Div. of Appl. Phys., Hokkaido Univ., Sapporo, Japan)

We present a new view point for effective moduli of acoustic metamaterials. We introduce the concept of hidden source of volume: the effective modulus is negative when the volume of fluid injected from the hidden source is larger than, and operates in antiphase to, the volume change of the unit cell that would be obtained in its absence. We demonstrate this ansatz for some established acoustic metamaterials with elements based on Helmholtz resonators and side holes. The hidden source for a Helmholtz-resonator-based metamaterial is the extra air volume injected from the resonator cavity. The hidden sources—more aptly termed hidden expanders of displacement in this case—can arise from light rigid trusses coupled to extra degrees of freedom for mechanical motion such as the case of coupling to masses that move at right angles to the wave-propagation direction.

1:30

4pPAa3. A broadband sound absorber based on lossy acoustic rainbow trapping metamaterials. Tuo Liu, Jie Zhu, and Li Cheng (Mech. Eng., The Hong Kong Polytechnic Univ., Rm. FJ610, 11 Yuk Choi Rd., Hung Hom, Kowloon, Hong Kong, toneliutuo@gmail.com)

Acoustic rainbow trapping (ART) metamaterials offer spatial-spectral modulation and intensive trapping of broadband sound, with strong acoustic dispersion that is absent in naturally occurring materials. However, thermal and viscous losses stemming from the acoustic boundary layers within narrow resonance regions of ART metamaterials has not been thoroughly studied and effectively utilized. Here, we would like to propose a lossy model of ART metamaterials. By taking advantage of the thermal and viscous losses, such lossy metamaterials can effectively work as a broadband sound absorber. A rigid surface perforated with very narrow subwavelength holes that can provide boundary-layer losses and have gradient depth along the propagation direction is constructed to mimic the lossy ART model. Full wave numerical simulation results indicate that the system's thermal and viscous losses are governed by the geometry of the resonance unit cells. Together with the intrinsic trapping effect, this lossy ART metamaterials function as a high performance broadband sound absorber. It may contribute to the optimization of sound absorption materials and wedge design of anechoic chambers.

1:45

4pPAa4. Evaluation of the resolution of a metamaterial acoustic leaky wave antenna. Jeffrey S. Rogers, Christina J. Naify (Acoust. Div., Naval Res. Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375, jeff.rogers@nrl.navy.mil), Matthew Guild (U.S. Naval Res. Lab., National Res. Associateship Program, Washington, DC), Charles A. Rohde, and Gregory J. Orris (Acoust. Div., Naval Res. Lab, Washington, DC)

Acoustic antennas have long been utilized to directionally steer acoustic waves in both air and water. Typically these antennas are comprised of arrays of active acoustic elements which are electronically phased to steer the acoustic profile in the desired direction. A new technology, known as an acoustic leaky wave antenna, has recently been shown to achieve directional steering of acoustic waves using a single active transducer coupled to a transmission line passive aperture. The leaky wave antenna steers acoustic energy by preferential coupling to an input frequency and can be designed to steer from backfire to endfire, including broadside. This paper provides an analysis of resolution as a function of both input frequency and antenna length. Additionally, the resolution is compared to that achieved using an array of active acoustic elements. [This work was supported by ONR.]

2:00

4pPAa5. Holographic metamaterial using an acoustic leaky wave antenna. Christina J. Naify, Matthew D. Guild, Theodore P. Martin, Charles A. Rohde, David C. Calvo, and Gregory J. Orris (Acoust., Naval Res. Lab, 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify.ctr@nrl.navy.mil)

Acoustic antennas in the form of active phased arrays have long been utilized to steer acoustic waves in both air and water. Although acceptable, these antennas form electrically complex, bulky structures. Acoustic leaky wave antennas have recently emerged as a method to achieve directional steering using a minimal number of active transducers coupled to an analog metamaterial aperture. In this study, a two-dimensional air-acoustic leaky wave antenna, coupled to only four sources, is shown to steer acoustic energy within a full three-dimensional space while also providing significant savings in terms of size, weight, and cost. By careful selection of the analog geometry, the aperture has the capability to produce a wide range of holographic acoustic images.

2:15

4pPAa6. Electromechanical hybrid metamaterial for the control of ultrasonic guided waves. Nesrine Kherraz, Lionel Haumesser, Franck Levassort (GREMAN UMR CNRS 7347, Univ. François Rabelais of Tours, 03 Rue de la Chocolaterie, Blois 41000, France, nesrine.kherraz@univ-tours.fr), Paul Benard, and Bruno Morvan (Laboratoire Ondes et Milieux Complexes, UMR CNRS 6294, Univ. of Le Havre, Le Havre, France)

Last years, a growing number of research has been devoted to the control of the behavior of Phononic crystals and Metamaterials through for instance tunability of the frequency position and/or width of a band gap. This can be achieved by using active materials such as piezoelectric materials. Here, a new class of hybrid metamaterials (HMMs) is proposed to tune the dispersion of guided Lamb waves. The studied HMM is made of a single homogeneous piezoelectric plate on which metallic electrodes are laid on. These electrodes are very thin in comparison to the plate thickness and allow changing locally the electric boundary conditions (EBCs). It is shown, experimentally and theoretically, that by simply modifying the EBCs, coupling between some modes can be induced, leading to the opening of a gap at the edge or within the first Brillouin zone. Moreover, a hybridization gap is observed for the first symmetrical Lamb modes, involving the electrical resonance associated to an inductive electrical circuit connected on electrodes. Such a system opens important perspectives for the development of SAW devices for radiofrequency applications.

2:30

4pPAa7. Experimental determination of pressure-dependent stiffness of a nonlinear acoustic metamaterial. Stephanie G. Konarski, Michael R. Haberman, Preston S. Wilson (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, skonarski@utexas.edu), Katia Bertoldi, Sahab Babaee (Harvard Univ., Cambridge, MA), and Jongmin Shim (Univ. at Buffalo, Buffalo, NY)

The field of acoustic metamaterials (AMM) involves the design of subwavelength structures to create material properties exceeding those of traditional composite materials. One such structure developed recently, known as a "Buckliball," displays strong pressure-dependent stiffness [Proc. Natl. Acad. Sci. USA **109**, 5978 (2012)]. The unique response of these structures results from element geometry: spherical elastomeric shells with thin, patterned circular membranes whose skeleton buckles when subjected to a differential pressure. We present measurements of the properties of a nonlinear effective medium consisting of Buckliballs immersed in water. The experimental apparatus consists of four elements: (1) a pressure vessel, which contains (2) a resonance tube filled with Buckliballs suspended in water, (3) an air-coupled ultrasonic system to monitor water height in the resonance tube, and (4) acoustic excitation and measurement instrumentation to obtain the frequency dependent acoustic response of the Buckliball and water-filled resonator. Estimates of the effective stiffness and density are obtained by measuring the fundamental resonance frequency and water column height as a function of overpressure. Experimental results are compared with model estimates and discussed in the context of nonlinear acoustic wave propagation. [Work partially supported by ONR.]

2:45

4pPAa8. Acoustic Scattering Cancellation: an Alternative to Coordinate Transformation Scattering Reduction. Theodore P. Martin, Charles A. Rohde (Code 7160, U.S. Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, theodore.martin@nrl.navy.mil), Matthew D. Guild, Christina J. Naify, David C. Calvo, and Gregory J. Orris (National Res. Council Res. Associate Program, Code 7160, U.S. Naval Res. Lab., Washington, DC)

Coordinate transformations have been extensively studied as a means to hide an object immersed in a fluid from an acoustic field. The coordinate transform approach relies on coatings with non-uniform and typically anisotropic material properties to guide an acoustic wave around the object. Acoustic scattering cancellation is an alternative approach that relies on modal cancellation within a desirable bandwidth to significantly reduce the acoustic scattering cross-section of an object. We present a direct comparison, using coatings with similar layer thicknesses and material properties, demonstrating that scattering cancellation achieves a similar level of scattering reduction compared to coordinate transforms, but with much thinner coatings. Furthermore, we present experimental results demonstrating significant reductions in the scattering cross-section of neutrally buoyant elastic cylinders suspended in an aqueous environment. Omnidirectional scattering reduction is obtained using a single isotropic coating through modal cancellation of the monopole and dipole modes of the cylinders. The modal cancellation is achieved by carefully tuning the coating's material properties and thickness such that the combined cylindrical object and coating exhibit effective medium properties identical to water. The scattering cancellation is close to 15 dB over a broad bandwidth for all reduced frequencies below $ka \cong 1$.

3:00–3:15 Break

3:15

4pPAa9. Iterative approaches to extend dilute homogenization predictions of Willis materials to higher volume fractions. Michael B. Muhlestein (Mech. Eng., Univ. of Texas at Austin, 3201 Duval Rd. #928, Austin, TX 78759, mimuhle@gmail.com) and Michael R. Haberman (Appl. Res. Labs. and Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Dynamic homogenization of generally heterogeneous elastic materials leads to three different effective material properties: stiffness, density, and parameters that couple momentum to strain and stress to velocity. This formulation was first proposed by Willis [Wave Mot. **3**, 1–11, (1981)] and is thus commonly referred to as Willis coupling. While there are some homogenization methods which account for Willis coupling, those models rely on either the assumption of periodic media or on dilute concentrations of coupled inhomogeneities in an otherwise homogeneous matrix. This work presents the use of dilute models with iterative micromechanical methods to predict general material properties in materials with no long-range order and elevated volume fractions of inhomogeneity. Two such methods, self-consistent and differential effective medium models, will be presented for the homogenization of composites of arbitrary anisotropy and Willis coupling resulting from the presence of Willis coupled inclusions in an elastic matrix material. Predictions from both models are presented and compared with similar predictions using homogenization schemes which do not account for Willis coupling. It is found that incorporating Willis coupling in these iterative approaches does not significantly affect the prediction of the tensors describing the effective stiffness and density of the mixture.

3:30

4pPAa10. Mie scattering obviates echoes in concert halls. James B. Lee (6016 S. E. Mitchell, Portland, OR 97206, cadwal@macforcego.com)

Mie-resonant-scattering from objects (of sizes similar to impinging waves) embedded in a plane surface has two aspects: it distributes sound into a pattern approximating a cosine law of diffuse reflection; it deconstructs phase relations among components comprising acoustic signals. The former is much better known than the latter. But in large rooms like concert halls, phase-preserving specular reflections constitute echoes, compromising musical information. Deconstructing phases of reflected musical signals by diffuse reflection confines temporal information to direct sound, presenting only the frequency spectrum in reflected sounds.

3:45

4pPAa11. Synthesis of wave surface by frame loudspeaker array with limited spatial frequency band. Akio Ando, Daisuke Yamabayashi, and Takuya Yamoto (Dept. of Electric and Electronics Eng., Faculty of Eng., Univ. of Toyama, 1-10-11 Kinuta Setagaya, Tokyo 157-8510, Japan, andio@eng.u-toyama.ac.jp)

We have studied a frame loudspeaker array for sound reproduction of three-dimensional television aiming at realizing clear localization and better sound depth control. Because such array has no secondary sources in its center region, the reproduced wave surface is sometimes deteriorated even if the least squares optimization will be used to reshape the surface. Over the past few decades, a considerable number of studies have been made on the sound field reproduction by loudspeaker array. Recently, higher order ambisonics (HOA) and near-field compensated HOA (NFC-HOA) attract attention of the researchers. It is well known that these methods can limit a spatial frequency band by controlling the degree of spherical function. If such limitation is introduced to the frame loudspeaker array, the deterioration of wave surface will be alleviated because the optimization will be easy to progress by matching only the low spatial frequency components of the reproduced wave surface with those of the reference. The computer simulation showed that the least squares optimization with the moderate restriction of spatial frequency band brings better wave surfaces than that without the restriction.

4:00

4pPAa12. On the use of the Willis Constitutive Laws of Elastodynamics of Stratified Media in reflection/transmission problems. Olivier Poncelet and Alexander L. SHUVALOV (UMR CNRS I2M, Univ. of Bordeaux, I2M - Site Bat. A4, 351 Cours de la Liberation, TALENCE 33405, France, olivier.poncelet@u-bordeaux.fr)

The generalized constitutive relationships derived by Willis[1,2] for elastodynamics of composites have incited an increasing interest in the community of effective media, and particularly in that of metamaterials. The structures for which homogeneous and dispersive equivalent media are sought include laminates,[3,4] phononic crystals,[5] as well as locally resonant materials.[6] Those constitutive laws propose an extended vision of the notion of an effective medium since they fully generalize the linear relationship between the couple momentum/stress and the kinematic one particle velocity/strain through the tensors of anisotropic mass density (order 2), of elasticity (order 4) and of inertial coupling (order 3). Following general results obtained in [3] that provide the complete set of dispersive effective parameters describing exactly the "macroscopic" propagation in stratified media, this communication aims at exemplifying the use of the Willis model in different types of problems with interfaces coupling several media, at least one of which is actually inhomogeneous and described as a Willis effective medium. [1]G. W. Milton, J. R. Willis, Proc. R. Soc. A **463**, 855 (2007) [2]J. R. Willis, Mech. Materials **41**, 385 (2009) [3]A. L. Shuvalov *et al.*, Proc. R. Soc. A **467**, 1749 (2011) [4]S. Nemat-Nasser and A. Srivastava, J. Mech. Phys. Solids **59**, 1953 (2011) [5]A. N. Norris *et al.*, Proc. R. Soc. A **468**, 1629 (2012) [6]D. Torrent *et al.*, Phys. Rev. B **92**, 174110 (2015).

4:15

4pPAa13. A novel image based approach to evaluate the room impulse response. Ambika Bhatta, Charles Thompson, and Kavitha Chandra (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu)

In this work, a novel exact image based solution for a room impulse response is formulated. The resulting response of the rectangular room is an exact solution which is represented in terms of image-sources and branch integrals. It is shown to resolve some of the numerical and analytical constraints of the image based, geometric acoustics models and mode based solutions. The results are quantified in terms of the characteristic parameters, frequency and the dimensions of the room. The computation is also shown by using the features of parallel programming.

THURSDAY AFTERNOON, 26 MAY 2016 — SALON H, 1:00 P.M. TO 3:45 P.M.

Session 4pPAb

Physical Acoustics: Multiple Scattering II

Valerie Pinfield, Cochair
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Josh R. Gladden, Cochair
Physics & NCPA, University of Mississippi, 108 Lewis Hall, University, MS 38677

Contributed Papers

1:00

4pPAb1. Experimental validation of shear-mediated contributions to multiple scattering in concentrated random dispersions of spherical particles. Valerie J. Pinfield, Derek M. Forrester, and Jinrui Huang (Chemical Eng. Dept., Loughborough Univ., Loughborough LE11 3TU, United Kingdom, v.pinfield@lboro.ac.uk)

Multiple scattering models for ultrasound propagation in dispersions of spherical particles have conventionally included only multiple scattering of the compressional wave mode. Recent developments to these models by Luppé *et al.* [J. Acoust. Soc. Am. **131**, 1113 (2012)] have incorporated the effects of mode conversions into the multiple scattering model; these arise from shear and thermal wave modes produced by scattering at each particle. In our recent work, we have reported the identification of the dominant contributions to effective attenuation in such dispersed systems for either solid or liquid particles, and have reported both analytical and numerical solutions for them. Here, we present the key results for *shear*-mediated multiple scattering effects which are dominant in concentrated systems of small *solid* particles (sub-micrometer) in the mega-Hertz frequency range. We show experimental validation of the model predictions for silica particles in the size range 100 nm to 1 micrometer and 1 to 20 MHz using two different spectroscopy techniques, first, a pseudo-continuous wave spectrometer (the Malvern Ultrasizer), and secondly a pseudo-random binary sequence cross-correlation spectrometer (Digusonic DSX) suitable for in-line process monitoring.

1:15

4pPAb2. Anomalous diffusion of multiply scattered ultrasound in a two-component disordered medium. Sébastien O. Kerhervé and John H. Page (Dept. of Phys. and Astronomy, Univ. of MB, Allen Bldg., Winnipeg, MB R3T 2N2, Canada, sebastien.kerherve@umanitoba.ca)

Measurements of multiply-scattered wave transport allow the characterization of heterogeneous media and may reveal anomalous wave properties. When a wave propagates through a sufficiently disordered medium, it will undergo many scattering events. This may lead to diffusive transport or

even, in highly disordered media, to localized behavior in which transport comes to a halt. These two regimes can be distinguished by measuring the evolution of the transverse intensity profile when point-like ultrasonic pulse is incident on the opposite side of a slab-shaped sample. In the case of diffusive propagation, the profile width grows without limit as the square root of time, while for localized waves, the width reaches a saturation value at long times due to the trapping of the waves in the medium. In our experiment, the sample consists of aluminum beads randomly packed in silicone oil. Contrary to expectations, the width goes through a maximum as a function of time and varies slowly afterwards. This novel behavior may be due to the existence of two coupled modes of propagation: a fast component travelling through the liquid and a slower component traveling through the bead network. The resonances of the beads influence strongly the propagation inside the solid network.

1:30

4pPAb3. Controlling the exceptional points of parity-time symmetric acoustics. Chengzhi Shi, Marc Dubois (Dept. of Mech. Eng., Univ. of California, Berkeley, 3112 Etcheverry Hall, Berkeley, Berkeley, CA 94720, chengzhi.shi@berkeley.edu), Yun Chen, Lei Cheng (Dept. of Microelectronics, Fudan Univ., Shanghai, China), Hamidreza Ramezani, Yuan Wang, and Xiang Zhang (Dept. of Mech. Eng., Univ. of California, Berkeley, Berkeley, CA)

Parity-time (PT) symmetric systems experience phase transition between PT "exact" and "broken" phases at exceptional point. These PT phase transitions contribute significantly to the design of single mode lasers, coherent perfect absorbers, isolators, diodes, etc. However, such exceptional points are extremely difficult to access in practice because of the dispersive behavior of most loss and gain materials required in PT symmetric systems. Here, we introduce a method to systematically tame these exceptional points and control PT phases. Our experimental demonstration hinges on an active acoustic element that realizes a complex-valued potential and simultaneously controls the multiple interference in the structure. The manipulation of exceptional points offers new routes to broaden applications for PT symmetric physics in acoustics, optics, microwaves, and electronics, which are essential for sensing, communication, and imaging.

1:45

4pPAb4. Acoustic metafluids with multiple negative-index bands. Thomas Brunet, Olivier Poncelet, Christophe Aristegui (UMR CNRS I2M, Univ. of Bordeaux, I2M - Site Bat. A4, 351 Cours de la Liberation, TALENCE 33405, France, olivier.poncelet@u-bordeaux.fr), Jacques LENG (UMR CNRS LOF, Univ. of Bordeaux, Pessac, France), and Olivier MONDAIN-MONVAL (UMR CNRS CRPP, Univ. of Bordeaux, Pessac, France)

The extraordinary properties of acoustic (random) metamaterials, such as negative refractive index, originate from low frequency resonances of sub-wavelength particles. While most of these functional materials are fabricated by mechanical engineering, we have recently shown that soft matter techniques coupled with microfluidics open a new synthesis route for acoustic metamaterials especially for ultrasonics [1]. As a demonstration, we have achieved 3D-bulk acoustic metafluids with an alternatively low, zero, and negative index by producing large amounts of calibrated soft porous microspheres, acting like strong Mie resonators [2]. The wide variety of physicochemical processes offered by chemical engineering allows for the tuning of the resonant particle properties over a broad range of mechanical/acoustical parameters. In this talk we show that, according to a fine control of the Poisson coefficient of the macro-porous resonators, it is not only possible to achieve soft acoustic metamaterial with one negative band, but also with two separate and tunable ones [3]. As both their width and depth depend on particle properties, the "soft approach" should benefit the design and fabrication of building-block composites with specific extreme properties required in some targeted applications, such as spatial control of sound for beamforming or cloaking. [1] Brunet *et al.*, *Science* **342**, 323–324 (2013). [2] Brunet *et al.*, *Nat. Mater.***14**, 384–388 (2015). [3] Raffy *et al.*, *Adv. Mater.* **DOI**: 10.1002/adma.201503524 (2016).

2:00

4pPAb5. Multi-layered elastic shell metamaterial elements for improved transmission in sonic crystals. Alexey S. Titovich (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd., Bethesda, MD 20817, alexey.titovich@navy.mil), Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., NB, NJ), and Stephen D. O'Regan (Naval Surface Warfare Ctr., Carderock Div., Bethesda, MD)

A metamaterial-based sonic crystal, comprised of a lattice arrangement of cylindrical elastic shells, can be tuned to provide wave steering or phononic band filtering. The effective acoustic properties of such sonic crystals are determined by the geometric and material properties of each individual shell and the lattice spacing between shells. By matching the effective acoustic properties to those of the surrounding fluid, the sonic crystal can be made acoustically transparent at low frequencies, leading to improved transmission over that of non-matched shells. Bi-layered shells provide additional design parameters to broaden the region of acoustic transparency provided by a simple shell. In this effort, three-layered shells are employed to improve transmission further. Transformation acoustics analysis is applied to determine theoretical density and bulk modulus distributions, and then real-world materials are selected which best match these theoretical distributions. Finally, shell geometry is optimized to minimize scattering while constraining the effective acoustic properties to match those of the external fluid.

2:15

4pPAb6. Disk cavities in soft materials: Their bubble-like resonance and use in thin underwater sound blocking materials. David C. Calvo and Abel L. Thangawng (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, david.calvo@nrl.navy.mil)

The acoustics of gas-filled cavities in soft viscoelastic solids, such as rubber and gels, has been a renewed subject of research in recent years owing to usefulness in studies of multiple scattering and importance to sound insulating and anechoic materials. A brief review is followed by a presentation of recent research on disk cavity resonance done at the Naval Research Laboratory [J. Acoust. Soc. Am. **138**, 2537–2547]. A lumped parameter analysis of the breathing mode of a disk cavity is presented which yields a natural frequency expression valid for a high-aspect ratio cavity embedded in an elastic medium. A verification approach using finite-element methods is also described which directly computes resonance in the framework of COMSOL Multiphysics. Calculation of scattering cross sections and visualization of the elastic displacement field indicates the importance of shear wave radiation. As an application example, a specially designed single layer array of disk cavities in a thin silicone rubber (PDMS) sheet was modeled that resonantly blocks underwater sound by nearly 20 dB for a favorable wavelength/thickness ratio of 240. Disk cavities are found to provide a wider bandwidth than near-spherical cavities. [Work sponsored by the Office of Naval Research.]

2:30–2:45 Break

2:45

4pPAb7. An investigation of free flooding, air-filled underwater resonators. Andrew R. McNeese, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark S. Wochner (AdBm Technologies, Austin, TX)

This paper investigates the acoustic behavior of free flooding resonators consisting of inverted cups that are lowered into a water column such that a volume of air is entrapped within the cavity upon submersion. Similar to a two-fluid Helmholtz resonator, the resonance frequency is determined by the compliance of the entrapped air and the radiation mass of water at the opening of the cavity. The entrapped air mass is compressed as the resonator is lowered to depth due to the increase in hydrostatic pressure, which affects

the acoustic behavior. Collections of similar resonators with compliant walls were previously investigated for use in underwater noise abatement [Lee *et al*., Proc. Meeting Acoustics **22**, 045004 (2015)]; however, recent work has shown an increase in quality factor and attenuation performance for stiffer walled resonators. Measurements were taken to determine the resonance frequencies and quality factors of both individual resonators and arrays of resonators comprised of various wall materials as a function of water depth, resonator geometry, and spacing between resonators. Low frequency sound attenuation through arrays of resonators was also measured. These measurements and associated model comparisons will be discussed. [Work supported by AdBm Technologies.]

3:00

4pPAb8. Acoustic time reversal in granular media. Maxime Harazi, Yougu Yang, Mathias Fink, Arnaud Tourin, and Xiaoping Jia (ESPCI ParisTech, PSL Res. Univ., CNRS, 1 rue jussieu, Paris 75005, France, maxime.harazi@espci.fr)

In a non-dissipative medium, the wave equation is symmetric in time. Therefore, for every wave diverging from a pulsed source, there exists a wave that retraces all its original paths in a reverse order and converges at the original source. In the early nineties, M. Fink proposed a method for generating such a time-reversed wave. This method was first tested with ultrasound and then successfully extended to other types of waves such as microwaves, water waves, and even in optics. Several studies have shown that time reversal wave focusing is very robust to disorder. Here, we investigate time reversal (TR) of elastic waves propagating in fragile granular media consisting of glass beads under static compression. Pulsed elastic waves transmitted from a compression or a shear wave source are measured, time reversed, and back-propagated. The ability of the time-reversed wave to focus at the initial source is checked as a function of the source amplitude. We find that TR of the ballistic coherent wave is very robust to perturbations but provides poor resolution. By contrast, the short-wavelength scattered waves offer a finer TR focusing but are sensitive to rearrangements induced by the forward propagation wave itself: at large source amplitudes, time reversal focusing is broken, due to sound-induced rearrangements but without visible grain motion. Experimental results are confronted with predictions from a numerical model in which the propagation medium is modelled by a percolating network of masses interacting via linear springs.

3:15

4pPAb9. The emergence of spurious arrivals in Green's function extraction and passive imaging. Jie Li and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, San Diego, CA 92093, jil004@ucsd.edu)

Cross-correlations of ambient noise at two receivers can extract the two-point Green's function, given that the wave-field is spatially uniform. The presence of scatterers that act as uncorrelated secondary sources can destroy this condition. We retrieve the Green's function in one-sided noise with a scatterer in an aeroacoustic experiment, using a pair of microphones (separation 1 m) parallel to the beach which generate one-sided surf noise. The scatterer is a 20-cm-radius polyvinyl chloride pipe 2.5 m to the landside of the microphone pair. We cross correlate 400–2000 Hz noise to retrieve the Green's function. The results show that spurious scattered arrival emerges in the cross-correlation functions when the source distribution is limited. This causes the generalized optical theorem to break down, and thus, there is no guarantee that the spurious scattered arrival cancels. However, the spurious waves have a geometric interpretation, which are useful for inversion. By combining the travel times of the spurious and physical scattered waves, only a pair of receivers is sufficient to locate the scatterer passively.

3:30

4pPAb10. Reducing the pour point of crude oil by using of ultrasonic wave. Delong Xu, Weijun Lin, Jingjun Deng, Chao Li, Lixin Bai, and Yan Yang (Inst. of Acoust., Chinese Acad. of Science, Beijing 100000, China, xudelong@mail.ioa.ac.cn)

With the global economical development, high pour point crude oil (HPPCO) is getting more and more attentions. Due to its higher pour point and worse fluidity when it is cold, more difficulties are faced for its extraction. Ultrasonic wave technology is investigated to reduce the pour point of Sudan HPPCO in this paper. First, an ultrasonic horn is proposed and made. Then, after processing for 3 min by the horn, a decrease of more than 3 of the pour point of Sudan HPPCO is obtained. Furthermore, compared with that without ultrasonic processing, the percentage of higher molecular weight constituents becomes more, while that of heavier molecular weight component is less. The mechanism that the pour point of HPPCO can be changed through ultrasonic processing is analyzed and investigated by an experiment that three types of paraffin wax whose molecular weights are different are processed by ultrasonic wave. The main advantage of the ultrasonic processing for HPPCO is economical and environmental and the decrease of pour point after ultrasonic processing is irreversible.

Session 4pPP

Psychological and Physiological Acoustics and Speech Communication: Lessons from Interrupted Speech: Methods and Models

Valeriy Shafiro, Chair
Communication Disorders & Sciences, Rush University Medical Center, 600 S. Paulina Str., AAC 1012, Chicago, IL 60612

Chair's Introduction—1:30

Invited Papers

1:35

4pPP1. From macroscopic to microscopic glimpse-based models of intelligibility prediction. Martin Cooke (Ikerbasque, Lab de Fonetica, Facultad de Letras, UPV-EHU, Paseo de la Universidad, 5, Vitoria, Alava 01005, Spain, m.cooke@ikerbasque.org), Yan Tang (Univ. of Salford, Salford, United Kingdom), and Mate A. Toth (Univ. of the Basque Country, Vitoria, Spain)

Miller and Licklider's explorations of the intelligibility of temporally interrupted speech, and later studies extending their findings to the spectro-temporal plane, have shown how the twin factors of sparseness and redundancy confer a high degree of robustness on speech in noise. The current contribution addresses two questions. First, to what extent can quantitative estimates of supra-threshold unmasked speech account for average (macroscopic) intelligiblity across a range of speech styles and masking conditions? We examine how well glimpse-based objective intelligibility metrics predict listeners' speech recognition scores for natural and synthetic speech in the presence of stationary and fluctuating maskers, and demonstrate reduced correlations for competing sources with an informational masking component. The second question concerns which additional components, beyond speech glimpses, are required to make (microscopic) predictions of actual listener confusions at the level of individual noisy speech tokens. Using corpora of speech-in-noise misperceptions, we show that in many cases the source of listener confusions is the misallocation of information from the masker, suggesting that estimates of supra-threshold unmasked speech alone are insufficient to explain speech intelligibility in noise.

1:50

4pPP2. Recognition of interrupted words in isolation and in sentences. Gary R. Kidd and Larry E. Humes (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405, kidd@indiana.edu)

When listening to speech in the presence of competing sounds that render portions of a speech signal inaudible or unrecognizable, the ability to utilize partial speech information is crucial for speech understanding. To better understand this ability, we have been examining word recognition with different patterns of missing information (portions of speech replaced by silence) in words and sentences. Although we find that the major determinant of word recognition is the proportion of missing information, the location of glimpses and the pattern and predictability of glimpses can also influence performance. However, these effects depend on the presence or absence of a sentence context and the predictability of the target word within the sentence. With isolated words, the onset information is most important for recognition. For words in sentences, the importance of word onsets is diminished and depends on the amount of context provided by the sentence. When the pattern of interruptions throughout a sentence is manipulated, a predictable pattern of glimpses facilitates word recognition in a highly predictable sentence context, but not in a low-predictable context. The implications of these findings for theories of speech understanding under difficult listening conditions will be discussed. [Work supported by NIH (NIA and NIDCD).]

2:05

4pPP3. An information-theoretic approach to understanding interrupted speech. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Everyday listening conditions seldom present the listener with a completely intact speech signal. Research on understanding interrupted speech spans the last seven decades, yet few efforts have focused on what parts of the speech signal were being interrupted. Kewley-Port, Fogerty, and others explored the importance of consonants versus vowels for understanding speech when either was replaced by noise, but this approach is limited by their inherent differences such as segment duration. The present approach is inspired by Shannon information theory, where information is defined by unpredictability, uncertainty, or change. We developed a metric of biologically relevant spectral change in the speech signal, termed cochlea-scaled entropy (CSE). Sentences with high-CSE intervals (low predictability = high information) replaced by noise were understood more poorly than sentences with an equal number and duration of low-CSE intervals (high predictability = low information) replaced. CSE has been shown to better predict speech intelligibility than various temporal measures and consonant/vowel status. This approach has been validated across wide ranges of acoustic simulations of cochlear implant processing, suggesting that cochlear implant users might also utilize these information-bearing acoustic changes to understand speech. Extensions and future directions for this work will be discussed

2:20

4pPP4. Normal and Impaired hearing recognition of speech segments in noise. Jont B. Allen (ECE, Univ of IL, 1404 Sunny Acres Rd., Mahomet, IL 61853, jontallen@ieee.org) and Ali Abavisani (ECE, Univ. of IL, Urbana, IL)

The identification of very short phoneme segments of speech is the key to understanding speech in chopped noise. Over the last 12 years, UIUC has repeated Miller-Nicely's 1955 phone recognition experiment, with 60 subjects and six SNRs, from quiet to −22 dB SNR, providing an improve understanding of the fundamentals of phone perception in normal (NH) and impaired (HI) listeners. Our phone robustness metric (SNR50) is the SNR such that the phone error is 50% (Toscano and Allen (2014), JSHLR). The error rate at SNR50 + 5 [dB] is <0.33%. We interpret this to mean that above SNR50, phonemes are below the Shannon channel-capacity limit. This is a game-changer: We must reevaluate speech recognition methods. For example, in an experiment on HI ears, we found that HI ears make large errors (e.g., 100%) on a small subset of tokens (Trevino Allen, JASA **134**, 607, 2012). Averaging across tokens or listeners for any given consonant conflates the scores. There is a good news: Since there are only a small number of subject-dependent difficult sounds, testing time is reduced and accuracy is increased for a fixed test duration.

2:35

4pPP5. The contribution of amplitude modulations from speech and competing noise sources for speech recognition. Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, 1621 Greene St., Columbia, SC 29208, fogerty@sc.edu)

Speech is often heard in the presence of background noise during everyday listening conditions. Concurrent amplitude modulations of the speech and the interfering noise result in portions (i.e., "glimpses") of the speech signal that remain preserved at favorable signal-to-noise ratios while other portions of speech may be highly degraded. The acoustic and phonetic properties preserved during these glimpses play a role in determining the overall sentence intelligibility. Under such conditions, vocalic cues are relatively more preserved than consonantal/obstruent cues due to average differences in intensity. Investigations examining these different acoustic cues suggest important perceptual contributions from preserved speech amplitude modulations and non-simultaneous interactions from competing noise amplitude modulations. Results from several studies suggest that conditions that maximize the preservation of the sentence temporal envelope (e.g., during long glimpses or for vocalic intervals) result in higher levels of speech recognition. Amplitude modulation during noise-dominated intervals also impacts speech recognition performance. Overall, results suggest that the relative preservation and interaction (e.g., modulation masking or perceptual restoration) of amplitude modulations from the speech and noise is crucial for predicting speech recognition during noise interruption. [Work supported by NIH and ASHA.]

2:50–3:10 Break

3:10

4pPP6. Interrupted speech with competing talkers: Benefits of temporal envelope and periodicity cues for younger and older adults. William J. Bologna (Dept. of Hearing and Speech Sci., Univ. of Maryland, Medical Univ. of South Carolina, 135 Rutledge Ave., MSC 550, Charleston, SC 29412, bologna@musc.edu), Monita Chatterjee (Auditory Prostheses & Percept. Lab., Boys Town National Res. Hospital, Omaha, NE), and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Recognition of interrupted speech requires connecting speech fragments over time and across gaps of missing information. Intelligibility improves when silent intervals are filled with noise; the effect is enhanced when noise contains rudimentary speech information, such as the temporal envelope of the missing speech. In multiple-talker environments, recognition of interrupted speech is more difficult, particularly for older adults. In these cases, temporal envelope cues may provide an important scaffold for auditory object formation. Other basic speech cues, such as F0-related periodicity, may help listeners segregate multiple voices and provide additional benefit. The relative, and potentially additive, benefit of temporal envelope and periodicity cues, and their use by older adults, remain unclear. To address these questions, younger and older adults with normal hearing listened to sentences in quiet and competing talker backgrounds. Sentences were periodically interrupted with (1) silence, (2) envelope-modulated noise, (3) steady-state pulse trains, which contained periodicity information from the missing speech, or (4) envelope-modulated pulse trains, which provided both envelope and periodicity information. Results are discussed in terms of contributions of temporal envelope and periodicity cues to perceptual organization for younger and older adults listening in complex environments. [Work supported by NIH/NIDCD and a AAA Student Investigator Research Grant.]

3:25

4pPP7. Perceptual effects of fluctuating envelopes. Peggy B. Nelson (Univ. of Minnesota, 164 Pillsbury Dr. Se, Minneapolis, MN 55455, peggynelson@umn.edu) and Adam Svec (Starkey Hearing Technologies, Eden Prairie, MN)

Signal/masker envelope fluctuations have important effects on detection and discrimination. Narrowband Gaussian noise (GN) forward maskers yield higher masked thresholds for detecting pure tones than do low-fluctuation noise (LFN) forward maskers. The increased residual masking is due to inherent fluctuations in the temporal envelope of GN producing listener uncertainty. This uncertainty persists for longer durations for hearing-impaired (HI) than for normal-hearing (NH) listeners. In addition to listener uncertainty, amplitude-modulation (AM) forward masking may contribute to masking that occurs in complex listening tasks. In a recent study of AM forward masking, an unmodulated GN masker yielded more masking than an unmodulated LFN, suggesting that inherent envelope fluctuations were responsible for the amount of AM forward masking measured across listener groups. Contrary to predictions, there were no differences in AM forward masking between NH and HI listeners, revealing little effect of hearing loss on recovery from AM forward masking for this task. Considering the combination of listener uncertainty and AM forward masking, the persistence of masker envelope fluctuation effects likely lead HI listeners to experience sluggish recovery from prior rapid envelope fluctuations compared to NH listeners. Together, these findings may have implications for speech understanding in modulated or interrupted conditions. [Support: NIH-DC008306.]

3:40

4pPP8. Modulation masking and masked speech perception in normal-hearing school-age children and adults. Emily Buss, John H. Grose, Joseph W. Hall (UNC Chapel Hill, 170 Manning Dr., G190 Physicians, Chapel Hill, NC 27599, ebuss@med.unc.edu), and Christian Lorenzi (Ecole Normale Superieure & CNRS, Paris, France)

It has recently been suggested that adults' ability to recognize speech in noise is limited by the inherent modulation of that noise—a form of modulation masking. This study evaluated whether immature masked speech perception in school-age children could be due to greater susceptibility to modulation masking. To gauge modulation sensitivity, the first experiment measured sinusoidal modulation detection for rates of 10–300 Hz carried by a 5000-Hz pure tone. There were large individual differences, but little evidence of a child/adult difference. This adult-like modulation detection for a tonal carrier contrasts with published findings of adult/child differences in modulation detection for a noise-band carrier, suggesting that children may be more susceptible to modulation masking than adults. The second experiment evaluated masked sentence recognition for speech that was filtered into 28 adjacent equivalent rectangular bands (100–7800 Hz), with alternate bands presented to opposite ears. Maskers were composed of either noise bands or tones, one centered on each speech band. These stimuli have been argued to characterize effects of modulation masking. Young children tended to perform more poorly than adults overall. Masker effects will be discussed in terms of possible developmental differences in energetic and modulation masking.

3:55

4pPP9. Effects of adult aging on perception of alternated speech. Arthur Wingfield (Volen National Ctr. for Complex Systems, Brandeis Univ., MS 013, Waltham, MA 02453, wingfiel@brandeis.edu)

In 1954, Cherry and Taylor found that when presentation of a continuous speech message was rapidly alternated between the two ears, intelligibility progressively declined as the rate of alternation was increased up to 3–4 switching cycles/s (167–125 ms per ear). Intelligibility then improved as switching rates were increased beyond this point. This V-shaped function might occur if a finite time is required to switch attention from one ear to the other, during which time no usable information is available from either ear. Hence, the more frequently the speech is alternated between ears, the greater will be the cumulative loss of acoustic information. At alternation rates beyond 125 ms per ear the listener begins to adopt a strategy of attending to the interrupted signal from only one ear, relying on the redundancy of the large number of small speech segments to reconstruct the message (Miller and Licklider, 1950). We report data supporting the alternative view that the critical dimension underlying the point of minimal intelligibility is speech content per ear rather than time per ear. We then use this phenomenon as a test of whether adult aging results in a slowing of attentional shifts in audition.

4:10

4pPP10. Effect of language experience on the intelligibility of interrupted speech. Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu), Stanley Sheft (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL), Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX), and Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL)

Signal distortions are more detrimental to speech perception by native than non-native listeners. We investigated how speech clarity and semantic context influence the perception of interrupted speech. Native and non-native American English listeners heard semantically meaningful or anomalous sentences produced as conversational or "clear" speech, gated at different rates (0.5–16 Hz). Results showed that both semantic context and speech clarity had a significant rate-dependent impact on the intelligibility of interrupted speech. In general, intelligibility was higher for native than non-native listeners. However, the magnitude of the clear-speech benefit varied across the two listener groups. The clear-speech benefit was obtained for gating rates of 2 Hz and above, except for meaningful sentences with native listeners where the benefit begin at 1 Hz. Both listener groups were able to use contextual information but native listeners derived more benefit at lower gating rates, indicating greater ability to access semantic information with limited acoustic-phonetic input. The results suggest that the non-native speech-perception deficit in adverse listening conditions may in part be due to their less efficient use of compensatory information at higher levels of language processing.

4:25–4:45 Break

4:45

4pPP11. Tales from the dip: Factors in cross-rate intelligibility variation of interrupted speech. Valeriy Shafiro, Stanley Sheft, and Brendan Prendergast (Commun. Disord. & Sci., Rush Univ. Medical Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu)

Speech intelligibility involves integration of temporally and spectrally distributed acoustic information into higher-order perceptual categories to obtain individual words. In 1950, Miller and Licklider pioneered a simple but powerful method of interrupting speech that has been extensively used to investigate factors that make speech signals perceptually robust. Among numerous subsequent studies, a consistent finding has been a nonmonotonic relationship between intelligibility and interruption rate. As interruption rate increases, and the duration of speech fragments in each interruption cycle decreases, a U-shaped rate-intelligibility function with a dip around 1–5 Hz frequently emerges. While many factors (e.g., speech materials, task parameters, listener age, or hearing status) have been shown to influence performance at specific interruption rates, reasons for the appearance and location of the dip in the function have remained obscure. Previous work indicates that the location of the dip in the rate-intelligibility function can be altered predictably with changes in the temporal structure of the interrupted speech stream, and may vary with the duration of the corresponding perceptual units. These findings will be considered in the context of neurophysiological and information-processing models of interrupted speech, and used to suggest a framework to guide future research and practical applications.

5:00

4pPP12. The intelligibility of interrupted, time-compressed speech. Michelle R. Molis and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, michelle.molis@va.gov)

Speech is a highly redundant signal for young listeners with normal hearing, but access to that redundancy may be diminished for older listeners, with or without hearing loss. We compared the understanding of unprocessed speech with (1) time-compressed speech, (2) time-compressed speech expanded via interruptions, and (3) uncompressed speech interrupted with silence. Listeners were asked to identify the final four digits of spoken seven-digit strings presented in quiet and in a steady-state, speech-shaped background noise (SNR +5). Three uniform compression ratios (2:1, 3:1, and 5:1) were used. Participants were younger normally-hearing listeners (YNH), somewhat older listeners (mid-50s to mid-60s) well-matched to the YNH group in thresholds up to 4 kHz (ONH), and older listeners with moderate hearing impairment well-matched in age to the ONH group (OHI). Employing this method, we have demonstrated the relative importance of the amount of information contained between the interruptions; and, the influence of age and hearing loss on the effective use of that information. [Work supported by VA RR&D I01RX001020.]

5:15

4pPP13. Neuronal oscillations in decoding time-compressed speech. Oded Ghitza (Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215, oghitza@bu.edu)

At the core of oscillation-based models of speech perception is the notion that decoding is guided by parsing. In these models, parsing is executed by setting a time-varying, hierarchical window structure synchronized to the input. Syllabic parsing is into speech fragments that are multi-phone in duration, and it is realized by a theta oscillator capable of tracking the input syllabic rhythm, with the theta cycles aligned with intervocalic speech fragments termed theta-syllables. Prosodic parsing is into fragments that are multi-word in duration, and it is realized by a delta oscillator capable of tracking phrase-level prosodic information, with the delta cycles aligned with chunks. Intelligibility remains high as long as the oscillators are in sync with the input, and it sharply deteriorates once they are out of sync. In the pre-lexical layers, decoding is realized by a cascade of neuronal oscillators in the theta, beta, and gamma frequency bands, with theta as "master." This talk reviews a model that utilizes this cortical computation principle, capable of explaining counterintuitive data on the intelligibility of time-compressed speech hard to explain with conventional models of speech perception. [Work supported by AFOSR.]

5:30–5:50 Panel Discussion

Session 4pSC

Speech Communication and Signal Processing in Acoustics: Constraints, Strategies, and Economies of Effort in Speech Production: Joseph S. Perkell's Contribution to Speech Science

Jennell Vick, Cochair
Psychological Sciences, Case Western Reserve University, 11635 Euclid Ave., Cleveland, OH 44106

Frank Guenther, Cochair
Boston Univ., 677 Beacon Street, Boston, MA 02115

Lucie Menard, Cochair
Linguistics, Universite du Quebec a Montreal, CP 8888, Succ. Centre-Ville, Montreal, QC H3C 3P8, Canada

Chair's Introduction—1:30

Invited Papers

1:35

4pSC1. Sensory feedback control in speech: Neural circuits and individual differences. Frank H. Guenther (Boston Univ., 677 Beacon St., Boston, MA 02115, guenther@cns.bu.edu)

Speech production involves a combination of feedforward and sensory feedback-based control mechanisms. The latter have been characterized in experiments involving real-time perturbations during speech. Unexpected perturbations of auditory feedback result in corrective motor responses with a minimum latency of approximately 100 ms after perturbation onset. These responses have been shown for both pitch and formant frequency perturbations, and the responsible neural circuitry includes portions of the superior temporal gyrus and ventral premotor cortex (vPMC). Corrective responses to somatosensory perturbations (such as a downward force applied to the jaw) occur approximately 50 ms after perturbation onset and are mediated by ventral somatosensory cortex and vPMC. The degree to which an individual weights auditory versus somatosensory feedback varies substantially. Such differences rely in part on differences in sensory acuity, e.g., a speaker with relatively poor hearing is likely to rely more heavily on somatosensory feedback control mechanisms than auditory feedback control mechanisms. Additionally, reliance on feedback control may be modulated to compensate for impairments in the feedforward system for speech, e.g., adults who stutter show a reduced reliance on auditory feedback control compared to fluent speakers, perhaps because auditory feedback can have a deleterious effect on speech initiation in stuttering.

1:55

4pSC2. Sensorimotor development and speech production. Lucie Menard, Christine Turgeon, and Pamela Trudeau-Fisette (Linguist, Universite du PQ a Montreal, CP 8888, Succ. Centre-Ville, Montreal, QC H3C 3P8, Canada, menard.lucie@uqam.ca)

According to Perkell's view [J. Perkell, J. Neurol. **25**, 382–407], phonemic goals correspond to multidimensional spaces in the auditory and somatosensory dimensions. The relationships between sensory goals and motor actions in speech are acquired during infancy through feedback-based control mechanisms. Although there have been numerous studies of speech development, little is known about the specific roles of auditory and somatosensory feedback. This paper reviews a few studies of how pre-school-aged children rely on sensory feedback to reach phonemic targets. The studies looked at children with normal-hearing and deaf children with cochlear implants. Vowels and consonants were recorded in various prosodic contexts (neutral and under contrastive emphasis) and perturbation conditions (with or without artificial perturbations of the articulators). Acoustic, articulatory, and perceptual analyses were conducted to assess the degree to which children reached the targets. The data showed that although children's productions were more variable than adults, children generally adapted their articulatory gestures to produce the appropriate sensory speech goals. The effects of perceptual acuity and phonological knowledge in speech development are discussed.

2:15

4pSC3. Development of auditory and somatosensory goals for alveolar sibilant production in later childhood. Jennell C. Vick (Psychol. Sci., Case Western Reserve Univ., 11635 Euclid Ave., Cleveland, OH 44106, jennell@case.edu), Nolan T. Schreiber (Elec. Eng., Case Western Reserve Univ., Cleveland, OH), Rebecca L. Mental (Psychol. Sci., Case Western Reserve Univ., Cleveland, OH), Michelle L. Foye (Cleveland Hearing & Speech Ctr., Cleveland, OH), and Gregory S. Lee (Elec. Eng., Case Western Reserve Univ., Cleveland, OH)

An age-related increase in the fricative contrast /s/-/ʃ/, measured acoustically, occurs in children up to 7 years of age, with 5- and 7-year-olds producing contrasts that are greater than those of younger children, but still significantly smaller than the contrasts produced by adults. Kinematic studies have demonstrated that adult-like speech motor control does not emerge until later adolescence, although no articulatory studies of sibilant production in later childhood have been reported. In this study, the aim was to understand the relative use of somatosensory and acoustic/auditory goals for the production of the fricative contrast in older children (10–15 years of age) and adults. The fricative contrast was measured in the acoustic signal and in articulatory data gathered from the tongue blade. The acoustic and articulatory contrasts were analyzed to test the magnitude of the covariation of the two domains. We further analyzed the development of the contrast in both domains as a function of later speech development. Results will be discussed in the context of the hypothesis that sibilants are produced with prominent goals in both the somatosensory and auditory domains but that auditory goals predominate in older children because of continued refinement of feedforward commands.

2:35

4pSC4. Articulatory idiosyncrasy inferred from relative size and mobility of the tongue. Kiyoshi Honda (School of Comput. Sci. and Technol., Tianjin Univ., 135, Yaguan Rd., Jinnan Dist., Tianjin 300350, China, khonda@sannet.ne.jp), Honghao Bao, and Wenhuan Lu (School of Comput. Software, Tianjin Univ., Tianjin, China)

Origins of individual characteristics of speech sounds have been a mystery. Individual patterns of higher spectra could be attributed to quasi-static hypopharyngeal-cavity resonance, while those of lower spectra are puzzling because both spectra and vocal-tract shapes radically change during speech. A possible clue to look into articulatory idiosyncrasy may be the relation between relative size and mobility of the tongue in the oropharyngeal cavity. To this end, combined cine- and tagged-MRI collected from four Chinese speakers producing two-syllable words were processed. The relative tongue size was indexed by midsagittal tongue area divided by tongue plus airway area both measured above the level of the superior genial tubercle in static MRI during /i/. The mobility of the tongue was measured by average velocity of tag points located along the oral and pharyngeal surface of the tongue. In the result, the velocity monotonically decreased with the relative tongue size, suggesting that the smaller the tongue the faster the movement according to a speaker-specific anatomical constraint derived from the space available for tongue articulation in vowel production. [Work supported by National NSF Programs (No. 61573254 and 61304250), and National One-Thousand Program (WQ20111200010) in China.]

Contributed Paper

2:55

4pSC5. Patterns of lingual articulation: A real-time three-dimensional ultrasound + palate study. Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, slulich@indiana.edu)

Joe Perkell's influence on the science of speech articulation has been enormous, beginning with the publication of his doctoral dissertation in 1969, which investigated the motion of speech articulators within the midsagittal plane. This presentation follows in the same spirit, extended to three dimensions. Three-dimensional (3D) tongue shapes—with aligned palate models—for a variety of speech sounds will be presented, followed by a principal components analysis of 3D tongue shape variability. The data were collected using a 3D/4D ultrasound system to image the tongue, and a 3D laser scanner to digitize palate impressions. Unlike most MRI data, these ultrasound recordings were made in upright sitting position during real-time speech. A limitation is the lack of information regarding the positions of the soft palate, the posterior pharyngeal wall, and the lips. The focus with 3D/4D ultrasound is therefore centered on the oral cavity and the anterior pharyngeal wall (i.e., the tongue root).

Invited Papers

3:10

4pSC6. Mind the gap: Electromagnetic articulometer observation of speech articulation in conversational turn-taking. Mark Tiede (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, tiede@haskins.yale.edu), Dolly Goldenberg (Linguist, Yale Univ., New Haven, CT), and Christine Mooshammer (Linguist, Humboldt-Universität, Berlin, Germany)

Joe Perkell pioneered the use of electromagnetic articulometry (EMA) for the observation and quantification of the kinematics of speech articulator movements. In the spirit of his research, we have extended these methods to EMA observation of two facing speakers interacting in conversation. The gaps or pauses between turns in speaking are known from acoustic measurement to be relatively short in duration, about 200 ms or the length of a syllable on average, and this gap duration has been shown to be consistent across widely diverse languages and cultures (Stivers *et al.*, 2009). However, because the cognitive latencies for producing a response are much longer than this interval its planning must occur during the incoming turn. Here we provide evidence for this planning from articulator movements that anticipate speech. Movements of sensors attached to the tongue, jaw and lips have been tracked for each of 12 speaker pairs. Gaps are measured as the difference between the acoustic end of one speaker's turn to the onset of aggregated articulator movement above a 20% peak velocity threshold for the respondent. Results show that the speech articulators typically assume an appropriate posture for initiation of a speaking turn well before the onset of speech.

3:30–3:50 Break

3:50

4pSC7. Speech biomechanics: What have we learned and modeled since Joseph Perkell's tongue model In 1974? Pascal Perrier (Gipsa-lab, CNRS & Université Grenoble Alpes, 11 Rue des Mathématiques, Saint Martin d'Hères F-38400, France, Pascal.Perrier@gipsa-lab.grenoble-inp.fr), Yohan Payan (TIMC-IMAG, CNRS & Université Grenoble Alpes, La Tronche, France), Mohammad A. Nazari (Mech. Eng., Univ. of Tehran, Tehran, Iran), Nicolas Hermant (TIMC-IMAG, CNRS & Université Grenoble Alpes, La Tronche, France), Pierre-Yves Rohan (Institut de Biomécanique Humaine Georges Charpak, Arts & Métiers ParisTech, Paris, France), Claudio Lobos (Departamento de Informatica, Universidad Tecnica Federico Santa Maria, Santiago, Chile), and Ahmad Bijar (TIMC-IMAG, CNRS & Université Grenoble Alpes, La Tronche, France)

With his "physiologically oriented, dynamic model of the tongue, Joseph Perkell introduced in 1974 a new methodological approach to understanding the "relationships among phonetic models and the properties and capabilities of the speech-production mechanism." This approach has guided a large part of our studies in the two last decades. In order to investigate how mechanical properties of the orofacial motor system constrain the degrees of freedom of speech articulation and contribute to shaping the speech signals exchanged between speakers and listeners, we, among other research groups, have developed increasingly more realistic 2D, and then 3D, finite element (FE) biomechanical models of the human vocal tract and face. After summarizing some of our modeling and simulation results that shed light on some basic characteristics of speech production, we present recent developments which aim to improve the realism of the models: evaluation of the links between the FE mesh structure (based either on tetrahedra, hexahedra, or mixed elements) and simulation accuracy; development of an active 3D element that simulates muscle mechanics and muscle force generation mechanisms; use of Diffusion Tensor Imaging to investigate muscle anatomy; design of an Atlas-based method (i.e., without manual image segmentation) for the automatic generation of subject-specific models.

4:10

4pSC8. Acoustic and kinematic estimates of laryngeal stiffness. Victoria S. McKenna, Elizabeth S. Heller Murray, Yu-An S. Lien, and Cara E. Stepp (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cstepp@bu.edu)

Vocal fold movements differ between individuals with typical voices and those with voice disorders associated with increased laryngeal stiffness. Modeling suggests that changes in vocal fold kinematics correspond to changes in laryngeal muscle stiffness. In this study, 12 healthy adults produced repetitions of /ifi/ while varying their self-perceived vocal effort during simultaneous acoustic and nasal-endoscopic recordings, in order to compare a kinematic estimate of laryngeal stiffness to an acoustic measure. The acoustic measure, relative fundamental frequency (RFF), was determined from the last ten voicing cycles before the voiceless obstruent (offset) and the first ten voicing cycles of the following vowel (onset). A kinematic stiffness ratio was calculated by normalizing the maximum angular velocity by the maximum value of the glottic angle during vocal fold adductory gestures. A linear mixed-effect model found that RFF accounted for 52% of the variance in the kinematic data. Examined within-subject, 83% of participants exhibited at least a moderate negative linear relationship ($r = -0.5$ to -0.91) between the offset cycle 10 RFF and the kinematic stiffness ratio. Overall, the relationship between the acoustic and kinematic measures was strong and both measures showed consistent changes during self-modulated changes in vocal effort.

4:30

4pSC9. Speaking one's mind: Vocal biomarkers of depression and Parkinson disease. Satrajit S. Ghosh (McGovern Inst. for Brain Res., MIT, 43 Vassar St., 46-4033F, Cambridge, MA 02139, satra@mit.edu), Gregory Ciccarelli (MIT Lincoln Lab., Cambridge, MA), Thomas F. Quatieri (MIT Lincoln Lab., Lexington, MA), and Arno Klein (Sage Bionetworks, Seattle, WA)

Mental health disorders affect one in four adult Americans and have a staggering impact on the economy. Improved assessment and early detection of mental health state can reduce this impact. Voice analysis has been linked to depression and other mental health disorders. However, difficulties in data collection, variation in collection methods, and computational demands of analysis methods have limited the use of voice in mental health assessment. The pervasive use of smartphones offers a unique opportunity to understand mental health symptoms and speech variation in large groups. We have developed open source mobile applications to collect data and developed voice-based algorithms for predicting mental health state in major depressive disorder and Parkinson disease. We demonstrate the use of biophysical speech production models in creating and improving features for machine learning, in contrast to traditional approaches for feature extraction. A model-based approach fuses prior knowledge of the system with the input data, constrains the space of parameters to biophysically realistic values, and reduces overall prediction error. This joint work with MIT Lincoln Laboratory and Sage Bionetworks couples mobile sensors to effective feature extraction and prediction models to enable a scalable approach for estimating individual variation in mental health disorders.

4:50

4pSC10. Objective assessment of vocal hyperfunction. Robert E. Hillman, Daryush Mehta (Voice Ctr., Massachusetts General Hospital, One Bowdoin Square, Boston, MA 02114, hillman.robert@mgh.harvard.edu), Cara Stepp (Boston Univ., Boston, MA), Jarrad Van Stan (Voice Ctr., Massachusetts General Hospital, Boston, MA), and Matias Zanartu (Universidad Tecnica Federico Santa Marıa, Valparaiso, Chile)

Vocal hyperfunction (VH) is associated with the most frequently occurring types of voice disorders including benign vocal fold lesions (e.g., nodules) and dysphonia that occurs in the absence of concurrent pathology (e.g., muscle tension dysphonia). In 1989, Hillman *et al*. (J. Speech Hear. Res. **32**, 373–392, 1989) proposed and initially validated the first experimental framework for hyperfunctional voice disorders that was based on using objective measures of vocal function to test basic concepts about VH. This initial work set the stage for a program of research aimed at improving the prevention, diagnosis, and treatment of hyperfunctional voice disorders by attaining a better understanding of the etiological and pathophysiological mechanisms that underlie specific disorders within the broad

range of those associated with VH. An update on the progress of this ongoing work will be provided including new insights gained through the use of ambulatory voice monitoring, machine learning, and computer modeling of phonatory mechanisms. [Work supported by NIH-NIDCD R33 DC011588 and the Voice Health Institute.]

5:10

4pSC11. Collaboration and results in research on speech motor control. Joseph S. Perkell (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA 02139, perkell@mit.edu)

Research on speech motor control was conducted in the Speech Communication Group in R.L.E. at M.I.T. between 1965 and 2012. Virtually all of this work was collaborative—with participation of group members and colleagues from other institutions in the United States and abroad. The work dealt with various aspects of speech production, including properties of the production and perception mechanisms and their influences on observed behaviors and control strategies. Examples will be presented that illustrate the indispensable roles of collaborators in studies of motor equivalence, relations between production and perception and the roles of feedback and feedforward control. Collaborators' contributions were based on their expertise in a number of areas, including: electrical and biomedical engineering, experimental psychology, acoustic phonetics, speech perception, statistical analysis, neuro-computational modeling, and design of sophisticated hardware and software. These examples demonstrate how such collaborations were essential to a productive output of research findings and theoretical advances for over four decades. [Research supported by NINCDS and NIDCD, NIH.]

THURSDAY AFTERNOON, 26 MAY 2016 SNOWBIRD/BRIGHTON, 1:30 P.M. TO 5:05 P.M.

Session 4pSP

Signal Processing in Acoustics and Underwater Acoustics: Detection and Estimation in Uncertain Acoustic Environments II

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

Invited Papers

1:30

4pSP1. Bayesian model selection for a broadband coprime array with unknown number of broadband sources. Dane R. Bush and Ning Xiang (Architectural Acoust., Rensselaer Polytechnic Inst., 2609 15th St., Troy, NY 12180, danebush@gmail.com)

Coprime microphone arrays use sparse sensing to achieve O(MN) degrees of freedom using only O(M + N) elements, where M and N are coprime integers. The benefit is a narrow beam at frequencies higher than the spatial Nyquist limit allows, with residual side lobes arising from aliasing. These side lobes can be mitigated when observing broadband sources [D. Bush and N. Xiang, *J. Acoust. Soc. Am.,* **138**, 447–456 (**2015**)]. Peak positions indicate directions of arrival in this case; however, uncertainties on number of concurrent sound sources in practical applications challenge classical approaches to direction-of-arrival estimations. One has to first resolve the uncertainty on how many sources are present. In this work, Bayesian inference is used to first select which model the data prefer from competing models before estimating model parameters, including the particular directions of arrival. The model is a linear combination of modified Laplacian distributions (one per sound source). The posterior probability function is explored over the entire parameter space by nested sampling in order to evaluate Bayesian evidence for each model. Bayesian evidence is crucial for resolving the uncertainties regarding number of sources, preferring simpler models while penalizing unnecessarily complicated models in an inherent implementation of Occam's razor.

1:50

4pSP2. Bayesian localization, tracking, and environmental inference. Stan E. Dosso (School of Earth & Ocean Sci., Univ. of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper presents an overview of a general Bayesian inference approach to source localization, tracking and/or environmental estimation. Source location and spectral parameters together with environmental parameters are all considered unknown random variables to be estimated from prior information and observed acoustic data. The relative level of prior information for various parameters differentiates applications of interest. For instance, controlled-source geoacoustic inversion typically involves large prior uncertainties for seabed parameters but small uncertainties for source locations, although some applications, such as inverting noise from ships-of-opportunity, may involve larger location uncertainties. Alternatively, source localization in an uncertain environment typically involves non-informative location priors and environmental priors that reflect available knowledge. Tracking applications include additional prior constraints on source speed. In all cases, the goal is to compute marginals of the posterior probability density for source and environmental parameters, quantifying the information content of the data and prior. This is typically carried out with Markov-chain Monte Carlo methods including Metropolis-Hastings sampling and/or Gibbs sampling, with various approaches applied to improve efficiency (e.g., principal-component sampling, parallel tempering) and generality (trans-dimensional inversion). Multiple-source localization minimizes the Bayesian information criterion to estimate the number of sources.

2:10

4pSP3. Broadband cross-correlation processing—Taking advantage of high-frequency impulse response envelope at low frequency. Paul Hursky (HLS Res. Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

The impulse response in a refracting ocean with multipath can readily be modeled using ray tracing (using a high-frequency approximation) or measured directly using acomms systems (using high-frequency broadband waveforms). Forming the envelope of this impulse response results in a robust characterization of the channel, based on time differences and angles of arrival, that are well-resolved by virtue of the bandwidth and short wavelength at high frequency. Can such a characterization be used as an aid to processing lower frequency signatures that are narrower in bandwidth, and thus not as well resolved in angle, and more prone to fading due to multi-path mutual interference? Paths that interact with a layered seabed will introduce frequency-dependent effects that must be accounted for. We will present results of processing data from several vertical line receive arrays, where we have a variety of broadband and narrowband waveforms being transmitted from several towed sources, and can leverage the high-frequency transmissions as an aid to processing the low-frequency signatures.

2:30

4pSP4. Source symbol decisions in the presence of space and time varying shallow water acoustic response functions. Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, pgendron@umassd.edu), Kari Cannon, and Graham Entwistle (ECE Dept., Univ. of Massachusetts Dartmouth, Darmouth, MA)

Effective underwater acoustic communications requires source symbol decisions in the presence of an uncertain space and time varying acoustic response function. A hierarchical mixture Gaussian model is useful for modeling both the sparsity of arrivals as well as their spread in angle, Doppler and propagation delay [Canadian Acoust. **40**]. In this framework, the delay spread, the degree of sparsity, and the Doppler spread all must be marginalized based on the observed data. We discuss the degree of sparsity as well as the correlation among multi-path arrival times and how these uncertain features in the response function can be efficiently treated in a computationally reasonable and statistically efficient manner via the hierarchy. The approach relies on iterative coherent symbol decisions with an empirical Bayes approach to estimating the hyper-parameters of the model permiting flexibility to adapt to environmental conditions. It is shown that coherent multi-path combining and Doppler compensation are possible at extremely low signal to noise ratios (i.e., < -18 dB), at ranges in excess of 1 km and with throughputs exceeding 100 bps with single element reception. Results are shown for large bandwidth M-ary orthogonal sequences tailored only to a maximum allowable multipath spread.

Contributed Papers

2:50

4pSP5. Adaptive pulse compression for clutter mitigation in midfrequency active sonar. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com), Jonthan Botts (Appl. Res. in Acoust. LLC, Culpeper, VA), Charles F. Gaumond (Appl. Res. in Acoust. LLC, Washington, DC), and Ian Cummings (Appl. Res. in Acoust. LLC, Culpeper, VA)

Midfrequency active sonar can achieve a combination of fine range resolution and good signal-to-noise ratio (SNR) by transmitting low-crest-factor frequency-modulated (FM) waveforms that are pulse compressed by match filtering. While this linear filter optimizes SNR for signals in additive Gaussian noise, it has large-amplitude range sidelobes that can allow strong sources of clutter, such as fish schools, to mask weaker nearby targets. To address range sidelobes we consider the reiterative minimum mean-square error (RMMSE) adaptive-pulse-compression algorithm [Blunt and Gerlach, *Proc. IEEE Intl. Radar Conf.*, Sept. 2003]. RMMSE performance is known to degrade for small Doppler shifts in the received waveform (e.g., due to clutter internal motion), which, depending on the form of the FM sweep, can be partially mitigated by covariance-matrix tapers [Cuprak, M.S. Thesis, George Mason University, 2013]. We note that RMMSE is analogous to the minimum-variance distortionless-response (MVDR) beamformer: for each range cell it steers nulls the location of strong returns at nearby range samples. Motivated by this similarity and expectation of target sparsity in range, we extend the compressive beamforming approach developed in prior work to learning a compressive-sensing match filter. Preliminary results from this work are discussed. [Work supported by a NAVSEA Phase I SBIR award.]

3:05–3:20 Break

3:20

4pSP6. Inferences on target speed, depth, and range from a continuous wave transmission. Paul J. Gendron, Tamunoala Charles-Ogan (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., Darmouth, MA 02747, togan@umassd.edu), David C. Anchieta (ECE, Universidade Federal do Pará, Belém, Brazil), and Justin Conor (ECE Dept., Univ. of Massachusetts Dartmouth, Darmouth, MA)

Tracking an underwater mobile object by means of a continuous wave transmission is challenging in part due to the difficulty of drawing inferences on closely spaced tones associated with target depth and speed. Considered here is a bistatic sonar arrangement employed to infer the depth, speed and range of an oncoming submerged object. Computation of the full posterior probability distribution of the returned amplitudes and frequencies from both the prior distribution and a finite duration window of the received waveform is made by Markov-chain Monte Carlo sampling. A Gibbs sampler is employed to construct the posterior joint density of all parameters by taking full advantage of the analytic tractability of the conditional and marginal densities of the received amplitudes while those of the ordered frequencies are constructed numerically by either an inverse quantile sampling or a Metropolis-Hastings sampler. The inferred density of depth, range, and speed of the target is accomplished by constructing a numerical inverse-transformation of the forward propagation model.

3:35

4pSP7. Target localization in a reverberant shallow ocean waveguide with environmental uncertainty using a nonlinear frequency-difference signal processing technique. Brian M. Worthmann and David R. Dowling (Univ. of Michigan, 1231 Beal Ave., 2010 Lay Automotive Lab., Ann Arbor, MI 48109, bworthma@umich.edu)

Model-based signal processing for active sonar localization is typically infeasible due to insufficient knowledge of the acoustic environment. Additionally, in a shallow ocean environment, surface or bottom roughness create diffuse reverberation that can often obscure a desired target echo. Recently, a nonlinear signal processing technique was developed for passive acoustic source localization (*Worthmann, et al.,* 2015) that is applicable to high frequency sources in uncertain shallow ocean environments. This technique exploits a nonlinear field product (the autoproduct) and uses in-band hydrophone array measurements to determine field information in a lower, out-of-band, frequency regime where environmental uncertainties are less detrimental. When extended to monostatic active sonar with a vertical array, this technique allows a model-based signal processing algorithm to combat the detrimental effects of reverberation. The nonlinear signal processing algorithm is presented, along with simulations in a 5-km range, 200-m deep ideal waveguide with environmental uncertainties and significant reverberation at frequencies between 2- and 5-kHz. Successful detection and localization of a mid-water-column target is found to be possible at simulated signal-to-reverberation levels as low as −5 dB. Comparisons to existing signal processing detection and localization algorithms are provided. [Sponsored by the Office of Naval Research and the National Science Foundation.]

3:50

4pSP8. Detection of an object bottoming at seabed through modeling consecutive reflected signals. Sunho Kim, Sungbin Im (School of Electron. Engr., Soongsil Univ., 369, Sangdo-Ro, Dongjak-Gu, Seoul 06978, South Korea, sbi@ssu.ac.kr), Taebo Shim, and Steve G. Kargl (APL-UW, Seattle, WA)

In various fields, it is very important to detect an object bottoming at seabed. In this paper, a scheme is proposed for detecting an object bottoming at seabed in the shallow water. In this work, a monostatic active sonar system is considered to use a linear frequency modulated signal as a transmitted signal, and to send consecutive pings in the shallow water. An object is assumed to be randomly located within 1000 m from the transmitter. A reflected signal received at the receiver can be modeled using a previously received reflected signal with Wiener filtering, to produce a FIR filter with estimated coefficients. This filter can be applied to predict the next received signal with the current received signal as input. The energy can be computed from the error signal between the predicted and received signals. In the case when the object is not located in the search area, the reflected signal is very similar to each other, and thus the energies of the error signals are very small. However, in the case when the object is placed and the reflected signal is scattered from the object, the energy of the error signal may be increased because the coherence between the predicted and received signals is low. The advantage of this approach is not to require a prior information on the characteristics of the sediment with the assumption that the seabed is consisting of an uniform sediment. The verification of the proposed scheme is investigated in terms of detection probability and detection range through computer simulation.

4:05

4pSP9. Guided wave reconstruction in complex geometries with a dictionary learning framework. K. Supreet Alguri and Joel B. Harley (Elec. and Comput. Eng., Univ. of Utah, 40 South 900 East, apt 2G, Salt Lake City, UT 84102, kssupreet@gmail.com)

Guided waves are an attractive tool for structural health monitoring (SHM) due to their ability to interrogate large areas of a structure. Yet, guided waves are characterized by multi-modal and frequency dispersive behavior and quickly grow in complexity with the structure. For example, guided wave reflections from plate edges, fasteners, or joints will lead to complex, difficult to analyze data. Most SHM algorithms try to remove or ignore these reflections. Yet, knowledge about the reflections and their acoustic behavior can significantly improve detection and localization algorithms. Hence, accurate knowledge about guided waves reflections is of a significant interest in SHM. In this paper, we reconstruct and predict guided wave measurements from geometric environments with reflections. We leverage dictionary learning and sparse recovery algorithms to achieve this goal. Dictionary learning is used to learn the building blocks of guided waves from simulation data. Sparse recovery algorithms are used to create predictive models of wave propagation based on experimental data. From simulation results, we show that our framework can successfully predict wave propagation, including reflections, across an aluminum plate with an accuracy of more than 95% from just 20 measurements. We demonstrate similar results with experimental data.

4:20

4pSP10. Design and tests of an acoustoelectric logging tool. Junqiang Lu, Xiaodong JU, Honglin Zhao, Baiyong Men, Wenxing Duan, and Wenxiao Qiao (China Univ. of Petroleum-Beijing, 18# Fuxue Rd. Changping District, Beijing 102249, China, lujq@cup.edu.cn)

The acoustoelectric logging is a developing method, and the logging method can interpret the seepage properties of the pore formation especially, such as permeability and so on. A new acoustoelectric logging tool is presented. The tool is composed of two acoustic transmitting transducers, three acoustic receiving transducers, two excitation electrodes, and four receiving electrodes. The transmitting transducers can work in a phased mode, and more radiant energy can be effectively generated. The acoustic radiant energy is a very important factor for the acoustoelectric logging tool, and measurements are conducted in an anechoic tank with dimensions of 5.0 m $\times$ 5.0 m $\times$ 4.0 m. The pulse width of excitation signals is 70μs, and the peak value is above 3000 V. The main frequency of receiving waves is 9.52 kHz, sound pressure is 47.2 kPa, and transmitting voltage response level is 147.5 dB. The tool is tested in the experimental wells and exploration wells. While the acoustic transmitting transducers radiate acoustic excitation signals, acoustoelectric signals from interfaces and concomitant acoustoelectric signals with acoustic waves are all acquired in clastic rock formations.

4:35

4pSP11. Bayes Factor test for the discrimination of a submerged mobile object from a continuous wave transmission. Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, North Dartmouth, MA), David C. Anchieta (ECE, Universidade Federal do Pará, Universidade Federal do Pará, Belém, Brazil, davidca102@gmail.com), Tamunoala Charles-Ogan, and Graham Entwistle (ECE Dept., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

Detection of an underwater mobile object by means of a continuous wave transmission is a challenging problem in part due to the difficulty of making inferences on closely spaced tones that provide clues regarding the objects' depth and speed. Considered here is a bistatic sonar arrangement to detect the presence of an oncoming submerged object. The optimal receiver computes the Bayes factor associated with the hypothesis test. The analytic intractability of the marginalization associated with the composite nature of the hypothesis leads to numerical methods of integration. The prior density on the target present scenario is constructed by an inverse image transformation through the forward propagation model. Computation of the Bayes factor is accomplished by Markov-chain Monte Carlo sampling with a Gibbs sampler. Analytically tractable conditional and marginal densities of the tone amplitudes are exploited while the conditional density of the ordered frequencies are constructed numerically by an inverse quantile sampler. Detection performance of a receiver against a mobile target are illustrated and discussed.

4:50

4pSP12. Merging models and data: Predictive modeling for guided wave. Joel B. Harley, K. Supreet Alguri (Dept. of Elec. and Comput. Eng., Univ. of Utah, 50 S Central Campus Dr. Rm. 3104, Salt Lake Cty, UT 84112-9249, joel.harley@utah.edu), and Alexander Douglass (Dept. of Mech. Eng., Univ. of Utah, Salt Lake Cty, UT)

New simulation tools for nondestructive evaluation (NDE) and structural health monitoring (SHM) are enabling predictive power that will quicken inspection accuracies, minimize inspection costs, reduce the need for data, and create pathways to new automatous inspection and monitoring methods. Simulations can be especially powerful tools for analyzing

complex geometries and modern anisotropic materials, such as carbon-fiber reinforced composites, where NDE and SHM theory and practice is still developing. Yet, truly predictive simulations are yet to be realized. Most simulations rarely match with experimental data unless the simulation is meticulously tuned. In this paper, we describe a framework for merging existing models with experimental data to create better predictive simulations of guided waves and other complex acoustic environments. We leverage tools and theory from compressive sensing, sparse inversion, and convex optimization. We focus on guided waves due to their significant complexity and their wide use in SHM and other acoustic applications. Our results achieve prediction accuracies of greater than 90% for guided waves in both isotropic and anisotropic environments. We also demonstrate how predictive models can be used for a variety of applications, including time of arrival estimation and temperature compensation.

THURSDAY AFTERNOON, 26 MAY 2016 SALON A, 1:30 P.M. TO 4:05 P.M.

Session 4pUW

Underwater Acoustics: Acoustic Propagation in the Ocean

Ying-Tsong Lin, Chair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543

Chair's Introduction—1:30

Contributed Papers

1:35

4pUW1. Practical estimates of the coherent leakage of sound from a mixed layer surface duct. Adrian D. Jones (Maritime Div., Defence Sci. and Technol. Group, P.O. Box 1500, Edinburgh, SA 5111, Australia, bear-jones@adam.com.au), Alec J. Duncan (Ctr. for Marine Sci. and Technol., Curtin Univ., Perth, WA, Australia), and Zhi Y. Zhang (Maritime Div., Defence Sci. and Technol. Group, Edinburgh, SA, Australia)

In a mixed-layer surface duct, acoustic transmission at frequencies below that required for duct trapping, as well as at frequencies moderately higher, is subject to coherent leakage of energy to the thermocline. The rate of coherent leakage with range is related to the imaginary part of the horizontal wavenumber of a mode, for which an iterative technique is commonly used for evaluations. In order to obtain alternative direct analytic expressions of the leakage rate, the analysis of Furry, described, for example, by Pederson and Gordon (JASA **47**, 304–326, 1970), has been revisited. Consideration has been given to both very large and very small ratios between sound speed gradients in the duct and below the duct, and ratios of sound speed gradients typical of those encountered as sea. The leakage rate for the first mode was found to well approximate the total signal leakage, hence the proposed analytical expressions relate to the first mode. Leakage values based on these expressions are compared against results from both a modal model and a wavenumber integration model, and a pre-existing approximate expression for leakage, for a number of surface ducted scenarios for frequencies relevant to the onset of duct trapping.

1:50

4pUW2. Acoustic influences of front width in a coastal curved-front model. Brendan J. DeCourcy (Mathematical Sci., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, decoub@rpi.edu), Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Analysis of an ocean shelf-slope front model [Y.-T. Lin and J. F. Lynch, J. Acoust. Soc. Am. **131**, EL1–EL7 (2012)] shows how acoustic quantities such as horizontal wavenumber depend on feature parameters [DeCourcy *et al*., J. Acoust. Soc. Am. **137**, 2421 (2015)]. The front model has a sharp curved interface between two isospeed regions in a coastal wedge. In this talk, the front is modeled more realistically with continuous sound speed variation over an interval of specified width between the two isospeed regions. The same approach is used as for the sharp front to obtain expressions for acoustic normal modes in the wedge. The solution comparisons between the sharp and continuous fronts are described, including the characterizations of modes as trapped, leaky, or transition. Simplified forms of the dispersion relation for the horizontal wavenumber eigenvalues are investigated. The objectives are to examine how including a continuous sound speed variation changes any conclusions about parameter dependence for the sharp front, and to determine sensitivity to model parameters such as front location, front width, slope angle, source frequency, and sound speed. [Work supported by the ONR.]

2:05

4pUW3. Coupled acoustic mode equations in the ocean waveguide with rough sea surface and elastic bottom. Andrey K. Morozov (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com) and John A. Colosi (Dept. of Oceanogr. Graduate School of Eng. and Appl. Sci., Naval Postgrad. School, Monterey, CA)

Sound wave scattering caused by rough surfaces has been the subject of investigation for many decades. The problem has no exact analytic solution and approximate approaches are needed. In the ocean, scattering of sound by perturbed ocean boundaries couples acoustic modes and increases attenuation. In this paper, approximate coupled mode equations are analyzed using a small perturbation approach. The analytic solution connecting mode shapes with coupling mode coefficients has been derived. It is shown that the coupled mode matrix can be approximated by a linear function of one parameter, which is the perturbation of the sea surface. The equations are applied for one-direction sound propagation with a rough sea surface. For this one parameter model, the coupled mode differential equation has an easy solution in the form of a matrix exponent and can be useful for many applications. The proposed method can be easily extended to the case where modes are coupling by uneven bottom topography including elastic properties for compressional and shear waves.

2:20

4pUW4. Examination of rough surface scattering from water/air interface using hybrid parabolic equation model. Mustafa Aslan (Dept. of Phys., Turkish Naval Acad., 1 University Circle, Monterey, CA 93943, maslan1@nps.edu) and Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA)

Many ocean acoustic propagation models assume an idealized pressure-release boundary at the ocean surface. This is easily accomplished numerically using finite element, finite difference, and split-step Fourier techniques. For models based on the split-step Fourier algorithm, rough surfaces can also be treated, but require additional complexity in the definitions of the field and propagator functions through a field transformation technique. For this reason, it may be advantageous to model a rough water/air interface in a manner analogous to the bottom treatment by extending the calculation into the air medium, thereby simplifying the definitions of the propagator functions. However, standard approaches to treat density discontinuities in split-step Fourier algorithms invoke smoothing functions, which have been shown to introduce phase errors in range. In this work, the hybrid split-step/finite-difference approach introduced by Yevick and Thomson (1996) is implemented in the Monterey-Miami Parabolic Equation (MMPE) model for both the water/sediment and water/air interfaces. Particular attention is paid to comparisons between the rough surface scattering results from the field transformation technique (applied to a pressure release surface) and the hybrid split-step/finite-difference approach (applied to the water/air interface).

2:35

4pUW5. Sound propagation from the Canadian Basin to the Chukchi shelf during Summer 2015. Mohsen Badiey, Justin Eickmeier, Andreas Muenchow (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu), Y T. Lin (Woods Hole Oceanographic Inst., Woods Hole, Delaware), Timothy Duda, John Kemp (Woods Hole Oceanographic Inst., Woods Hole, MA), Matthew Dzieciuch, and Peter Worcester (Scripps Inst. of Oceanogr., La Jolla, CA)

During the summer of 2015 a pilot experiment for the Canadian Basin Acoustic Propagation Experiment (CANAPE) was conducted between deep Arctic basin and shallow Chukchi shelf. A vertical line array (VLA) was deployed on the shelf (72.336 N, 157.449 W) at a depth of 161 m from July 26 to August 13, 2015. Sound sources were deployed from R/V Sikuliaq at locations ranging from 131 to 375 km from the VLA. M-Sequences centered at 250 Hz (bandwidth of 62.5 Hz) were transmitted from each location. Discrete shipboard CTD data were conducted at each transmission location and continuous CTD data were recorded along the VLA. Water column data suggests a sound channel located vertically between Pacific and Atlantic waters near the depth of Arctic halocline layer. They also show upward shoaling of lower halocline waters onto the shelf. This upwelling over the continental slope moves the sound channel from offshore to onshore near the bottom. The axis of this sound duct is 100 m below the surface and is about 100 m thick. Received acoustic signals on the shelf increase rapidly in intensity at time scales that range from minutes to days. In this paper we show the acoustic variations are due to the variable sound speed profile. Signal intensity, noise, and temporal variability for geotime scale of minutes to hours are reported. Numerical simulations are conducted to investigate the associated acoustic effects in data. [Work supported by ONR 322OA.]

2:50–3:05 Break

3:05

4pUW6. Numerical analysis of underwater sound propagation over the Chukchi Sea shelfbreak. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu), Mohsen Badiey (Univ. of Delaware, Newark, DE), Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Andreas Muenchow (Univ. of Delaware, Newark, DE), John Kemp (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), Justin Eickmeier (Univ. of Delaware, Newark, DE), Matthew Dzieciuch, and Peter Worcester (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA)

During the 2015 Canada Basin Acoustic Propagation Experiment (CANAPE) expedition on the southern edge of Canada basin (northern Chukchi Sea shelfbreak), shipboard suspended underwater sound sources were deployed to transmit acoustic signals to hydrophone arrays moored on the deep basin as well as the shallow shelf. In this paper, numerical simulations utilizing the Parabolic Equation method are conducted to provide physical insights into the variability of signals propagating over the shelfbreak and slope and recorded on a vertical array on the Chukchi Sea shelf. The numerical models simulate sound propagating over the slope via the Pacific Halocline duct, which is a water-borne vertical sound duct formed between the layers of Pacific summer water and Atlantic water. The source to receiver distance is about 130 km, and realistic variability is introduced in the numerical models. The previous studies reported in the literature have concluded that the shelfbreak circulation, specifically upwelling, and the submesoscale eddies spun off the shelfbreak jet are the two major causes of the temporal and spatial variability of the Pacific Summer Water Layer over the slope in the region. The numerical simulation study also emphasize on the scattering and attenuation effects caused by ice cover and roughness. A preliminary data-model comparison is made and discussed in this paper. [Work supported by the Office of Naval Research.]

3:20

4pUW7. Potential impacts of climate change on acoustic propagation in the Arctic. Timothy F. Duda, Lee E. Freitag (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu), Lori R. Adornato (SRI Int., St. Petersburg, FL), and Robert H. Byrne (College of Marine Sci., Univ. of South Florida, St. Petersburg, FL)

Some forecasts show surface pH in the Arctic dropping from 8.1 to 7.6 over the next 100 years. This substantial decrease may cause changes in acoustic transmission at frequencies where the pH-dependent borate absorption plays a role, below about 5 kHz. In many cases, upward refraction of sound in the near-isothermal Arctic waters causes ice or surface scattering effects to dominate transmission. However, recent observations in the Canada Basin show that 700 and 900-Hz sound can be fully ducted beneath the Pacific Summer Water, with no ice interaction, and be detectable over distances of a few hundred kilometers. In this situation, the received signal level is controlled largely by cylindrical spreading and absorption. Here, for a wide band of frequencies, the effects of probable pH reductions and reduced absorption are investigated using a few models of pH depth profiles. There is potential for increased signal levels of 5 dB or more for 200-km propagation if the duct waters have significantly reduced pH.

3:35

4pUW8. Implementation of moving magnet actuation in very low frequency underwater acoustic transduction. Brenton Wallin (Sensors & SONAR Systems Dept., Naval Undersea Warfare Ctr., 30 Summit Ave., Narragansett, RI 02882, brenton.wallin@navy.mil), Steven Crocker, and Jeffrey Szelag (Sensors & SONAR Systems Dept., Naval Undersea Warfare Ctr., Newport, RI)

The Naval Undersea Warfare Center's Underwater Sound Reference Division (USRD) is designing a very low frequency sound source to calibrate line arrays in the 1–100Hz band at their Leesburg Facility. For calibrations below 20 Hz, the low frequency cutoff of their J15-3 standard projector, the USRD currently uses a passive method where the sound source is the ambient noise in the spring. A limitation of this method is a low signal-to-noise ratio. By designing, building, and implementing the aforementioned sound source, the USRD looks to increase the signal-to-noise ratio of their recorded data, and consequently increase the reliability of measurements at very low frequencies. This presentation explores the idea of implementing moving magnet technology in a very low frequency transducer. Although moving coil projectors have been used since 1914 as low frequency sound sources, their acoustic output at very low frequencies is limited. The potential benefits of a moving magnet projector over moving coil projectors are discussed. Acoustic performance estimates for a moving magnet projector are calculated based on specifications of an off-the-shelf moving magnet actuator. Lastly, the presentation will cover current efforts being made to design an underwater housing for the projector.

3:50

4pUW9. Reducing numerical dispersion and reflections in finite element and finite difference simulation of acoustic wave propagation and scattering. Murthy Guddati (NC State Univ., 2501 Stinson Dr., NCSU-Civil Eng., Raleigh, NC 27695-7908, mnguddat@ncsu.edu) and Senganal Thirunavukkarasu (Enthought, Inc., Austin, TX)

Discretization methods such as the finite element method (FEM) and the finite difference method (FDM) suffer from two types of accuracy problems: spurious wave dispersion for long-range propagation and artificial reflections from discretization of perfectly matched layers that are often used to represent the unbounded exterior. In this talk, we focus on lower-order FEM and compact FDM stencils, and show that these errors can be significantly reduced using simple measures: using modified integration rules for FEM, and modified compact stencils for FDM. It turns out that these needed corrections for reducing dispersion and reflections are in opposite directions, indicating that reducing one error results in increasing the other, which is not desirable. In this talk, we introduce a special technique that leads to reduction of both dispersion and reflection errors, and illustrate its effectiveness through theoretical analysis and numerical experiments.

THURSDAY EVENING, 26 MAY 2016 — 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Thursday are as follows:

Committee	**Start Time**
Noise	Salon D
Speech Communication	Salon G
Underwater Acoustics	Salon J