Session 5aAB

Animal Bioacoustics: Animal Hearing and Vocalization

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

Contributed Papers

9:00

5aAB1. A comparison of acoustic and visual metrics of sperm whale longline depredation. Aaron Thode (SIO, UCSD, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Lauren Wild (Sitka Sound Sci. Ctr., Sitka, AK), Delphine Mathias (GIPSA Lab., Grenoble INP, St. Martin d'Hères, France), Janice Straley (Univ. of Alaska Southeast, Sitka, AK), and Chris Lunsford (Auke Bay Labs., NOAA, Juneau, AK)

Annual federal stock assessment surveys for Alaskan sablefish also attempt to measure sperm whale depredation by quantifying visual evidence of depredation, including lip remains and damaged fish. An alternate passive acoustic method for quantifying depredation was investigated during the 2011 and 2012 survey hauls. A combination of machine-aided and human analysis counted the number of distinct "creak" sounds detected on autonomous recorders deployed during the survey, emphasizing sounds that are followed by a period of silence ("creak-pauses"), a possible indication of prey capture. These raw counts were then adjusted for variations in background noise levels between deployments. For most locations, the noise-adjusted counts of "creak-pauses" were highly correlated with survey counts of lip remains during both years (2012: $r(10) = 0.89$, $p = 1e\text{-}3$; 2011: $r(39) = 0.72$, $p = 4e\text{-}3$) and somewhat correlated with observed sablefish damage in 2011 [$r(39) = 0.37$, $p = 0.03$], but uncorrelated with other species depredation. The acoustic depredation count was anywhere from 3% to 80% higher than the visual counts, depending on the survey year and assumptions employed. The observed correlation breaks down when three or more whales are present. The results suggest that passive acoustics can provide upper bounds on the bias of survey depredation monitoring efforts for moderate depredation levels.

9:15

5aAB2. Equal loudness contours and possible weighting functions for pinnipeds. Colleen Reichmuth (Inst. of Marine Sci., Long Marine Lab., Univ. of California, 1, 100 Shaffer Rd., Santa Cruz, CA 95060, coll@ucsc.edu)

The idea of developing frequency weighting functions for marine mammals has received considerable attention recently because such functions can determine the relevant bandwidth for noise exposure assessments, and because they take differences in auditory sensitivity between species into account when identifying acoustic risks. However, such weighting functions are difficult to establish for nonhumans as they rely on equal loudness relationships that are subjective. Equal auditory reaction times may serve as a proxy for equal loudness judgments. For this experiment, we measured frequency-specific latency-intensity (L-I) functions for one California sea lion and one harbor seal with tones that were +0, +2, +4, + 6, +10, +20, +30, and +40 dB re: sensation level (SL). The L-I plots were reliably fit with a power function to enable the determination of sound pressure levels corresponding to discrete latency values for each subject at each frequency. From these data, equal latency contours were drawn to describe differential auditory sensitivity as a function of frequency. The weighting functions derived from these contours are less conservative than the currently proposed "m"-weighting function for marine mammals, and may be more reliable than the alternative inverted audiogram approach. [Work supported by ONR.]

9:30

5aAB3. Psychophysical studies of hearing in sea otters (*Enhydra lutris*). Asila Ghoul and Colleen Reichmuth (Inst. of Marine Sci., Long Marine Lab., Univ. of California Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, asila@ucsc.edu)

The sensory biology of sea otters is of special interest, given their amphibious nature and their recent evolutionary transition from land to sea. However, little is known about the acoustic sense of sea otters, including sensitivity to airborne and underwater sound. In this study, we sought to obtain direct measures of auditory function. We trained an adult-male southern sea otter to participate in audiometric testing in an acoustic chamber and an acoustically mapped pool. We used a psychoacoustic method of limits to determine absolute auditory thresholds in air and under water across the hearing range. In addition to obtaining aerial and underwater audiograms, we also evaluated hearing in the presence of noise. The otter's aerial hearing closely resembled that of a sea lion, and showed reduced sensitivity to high-frequency (>22 kHz) and low-frequency (<2 kHz) sounds relative to terrestrial mustelids. Under water, hearing was less sensitive than sea lions and other pinnipeds, especially at frequencies below 1 kHz. Critical ratios were >10 dB above those measured in pinnipeds, indicating that sea otters are not especially well-adapted for extracting acoustic signals from noise. These data suggest that evolutionary changes in hearing are secondary to other adaptations for semi-aquatic living.

9:45

5aAB4. Explanation of the loudness and other features of cicada sounds. Derke R. Hughes (Sensors & Technol. Office, Naval Undersea Warfare Ctr., Newport, RI), Allan D. Pierce (P.O. Box 339, East Sandwich, MA 02537, adp@bu.edu), Richard A. Katz (Sensors & Technol. Office, Naval Undersea Warfare Ctr., Newport, RI), and Robert M. Koch (Chief Technol. Office, Naval Undersea Warfare Ctr., Newport, RI)

A quantitative explanation is given of features of noise emitted by cicadas (classed as the loudest of all insects). Microphone data shows sounds are emitted in a sequence of closely spaced tone bursts. Listeners do not perceive the individual pulses because of the finite integration time of the ear. The principal sound radiators are two platelets referred to as tymbals, which vibrate after being struck by ribs that have undergone buckling. The energy of each sound pulse is initially stored in tensed muscles and is initially released via buckling into the kinetic energy of ribs, which strike the tymbals in a manner similar to that of a drumstick striking a drum. The tymbals "ring" at a frequency controlled by the mass of the tymbals and the springiness of the air cavity within the abdomen of the cicada. The wavelengths of the radiated sound are much larger than the tymbal radii but comparable to the overall dimensions of the cicada. The detailed theory explains the radiation pattern of the sound radiation, the amplitude of the sound, the number of cycles in each pulse, the radiation damping of the tymbal vibrations, and why the cicada is such an efficient radiator of sound.

10:00

5aAB5. Temporal patterns in echolocation and cave use by Guam Swiftlets (*Aerodramus bartschi*) in native and introduced habitat. Andrew J. Titmus (Zoology, Univ. of Hawaii at Manoa, 1910 East-West Rd., University of Hawaii, Honolulu, HI 96822, ajtitmus@gmail.com), Alexis B. Rudd (Hawaii Inst. of Marine Biology, Kaneohe, HI), Kevin M. Brindock (Naval Base Guam, Santa Rita, Guam), Marc O. Lammers (Hawaii Inst. of Marine Biology, Honolulu, HI), and Whitlow Au (Hawaii Inst. of Marine Biology, Kaneohe, HI)

The Mariana Swiftlet (*Aerodramus bartschi*) is a federally listed endangered species of is native to Guam and the Marianas Islands. There is also a small, introduced population of Marianas Swiftlets on the island of Oahu, Hawaii. The nesting cave in Oahu is a small tunnel built for agricultural irrigation. Marianas swiftlets live in caves, which they navigate using echolocation clicks. Ecological Acoustical Recorders (EARs) were modified with a omni-directional microphone with a flat frequency response and −63 dB sensitivity for bird recordings. Data were recorded at a sample rate of 80,000 and a duty cycle of 30 s of recording every 5 min. BEARs (Bird EARs) were placed in swiftlet caves on Oahu, Hawaii, and Guam where they recorded for between five and fifteen days. Swiftlet clicks were detected using Ishmael's energy sum detector. Temporal patterns of clicking were analyzed and compared between the two sites and correlated with environmental data over the recording period to determine effects of sub-optimal nesting habitat and changed weather patterns on the Oahu population compared to the native population in Guam.

10:15

5aAB6. Dynamic encoding of sensory information in biomimetic sonar baffle. Mittu Pannala (Mech. Eng., Virginia Tech., ICTAS II, Bldg. 116 Washington St., Blacksbulg, VA 24061, mpannala@vt.edu), Naren Ramakrishnan (Comput. Sci., Virginia Tech., Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech., Blacksburg, VA)

The biosonar system of horseshoe bats stands through several dynamic features that could be related to an outstanding sensory performance. The outer ears (pinnae) of the animals, for example, can change their shapes in a non-rigid fashion and thereby produce qualitative beampattern alterations. Such changes can be reproduced qualitatively with a highly simplified biomimetic baffle prototype. Such a biomimetic prototype is a crucial platform for measuring large amounts of data on the time-variant device behavior that would be difficult to obtain from an animal *in vivo*. Here, such time-variant data were used to investigate the question whether a dynamic deforming baffle can enhance the encoding of sensory information. This was done based on estimates of the mutual information between beampatterns belonging to different deformation stages of the biomimetic baffle. To make this estimation problem tractable, the transfer functions associated with each direction of sound incidence were sorted into "alphabets" of a small number (X-Y) of discrete signal classes using spectral clustering algorithms. Mutual information estimates computed based on these signal classes indicated very low dependencies between the beampatterns even from moderately distant points in the deformation cycle and hence support the notion of dynamic sensory encoding.

10:30

5aAB7. Static and dynamic control of emission beamwidth in horseshoe bats. Anupam Kumar Gupta (ME, Virginia Tech., 1208 Snyder Ln., Apt. #1900F, Blacksburg, VA 24060, anupamkg@vt.edu), Weikei He (School of Phys., Shandong Univ., Jinan, China), Dane Webster (School of Visual Arts, Virginia Tech., Blacksburg, VA), and Rolf Müller (ME, Virginia Tech., Blacksburg, VA)

Horseshoe bats (family Rhinolophidae) emit their ultrasonic biosonar pulses nasally with their nostrils surrounded by baffle structures known as "noseleaves." The noseleaves are often characterized by intricate local shape details such as flaps, ridges, or furrows. Furthermore, some of these structures are capable of undergoing non-rigid shape changes over time. One part of the horseshoe bat noseleaf that has both static local shape features as well as a time dynamics is the lancet, a vertical projection on the top of the noseleaf. The most striking static shape features of the lancet are half-open cavities (furrows) and the most obvious *in-vivo* motion of the lancet is a rotation in the distal-proximal about the base of the lancet. In the present work, the acoustic effects of the furrows and the distal-proximal rotation of the lancet were studied in three individuals of the Greater Horseshoe bat (*Rhinolophus ferrumequinum*) using numerical methods. The lancet furrows were always found to act as beam-focusing devices irrespective of lancet rotation. The acoustic effect of forward lancet rotation within the range seen in bats (tested angles 0°, 5°, 10°) was always a significant widening of the beam.

Session 5aBAa

Biomedical Acoustics and Physical Acoustics: Field Characterization and Dosimetry for Therapeutic Ultrasound Applications II

Vera A. Khokhlova, Cochair
Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Gail ter Haar, Cochair
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Chair's Introduction—7:55

Invited Papers

8:00

5aBAa1. Development and application of spot-poled membrane hydrophones for measurements of high intensity therapeutic ultrasound fields. Volker Wilkens (Ultrason. Working Group, Physikalisch-Technische Bundesanstalt, Bundesallee 100, Braunschweig 38116, Germany, volker.wilkens@ptb.de), Sven Sonntag (Gesellschaft für Angewandte Medizinische Physik und Technik, Merseburg, Germany), and Olga V. Georg (Ultrason. Working Group, Physikalisch-Technische Bundesanstalt, Braunschweig, Germany)

The reliable characterization of high intensity therapeutic ultrasound (HITU, HIFU) fields is important regarding the safe and effective clinical application of the modality. However, the required acoustic output measurements pose several metrological challenges. Extreme pressure amplitudes easily cause damage to the typical sensors, pressure waveforms comprise a large number of higher harmonics, and a small sensing element size is desirable due to the strong focusing. Membrane hydrophones are widely used as reference sensors due to their advantageous and predictable characteristics. However, they are usually considered to be rather fragile instruments possibly not well suited for HITU field characterization. A membrane hydrophone previously developed at PTB was tested by means of successive measurements at focus with increasing driving voltage, and the pressure range detectable without destruction of the hydrophone was determined. Second, a novel hydrophone design comprising additional protective layers and a backing was developed to increase the robustness against cavitation. After calibration, measurements were performed using an HITU transducer with working frequencies of 1.06 and 3.2 MHz. The examples show the favorable applicability for HITU field characterization. The maximum detectable rarefactional and compressional pressure amplitudes were 15 and 77 MPa, respectively, with a detection bandwidth of 50 MHz.

8:20

5aBAa2. Ultrasound velocity mapping with Lorentz force hydrophone. Pol Grasland-Mongrain (LabTAU, INSERM U1032, 151 Cours Albert Thomas, Lyon 69424, France, cyril.lafon@inserm.fr), Bruno Gilles (Université de Lyon, Lyon, France), Jean-Martial Mari (Imperial College, London, France), Benjamin Roussel, Adrien Poizat, Jean-Yves Chapelon, and Cyril Lafon (LabTAU, INSERM U1032, Lyon, France)

In previous work [Grasland-Mongrain *et al.* 2013], a Lorentz force hydrophone consisting of a cylindrical arrangement of magnets around a thin metallic wire has been presented. An ultrasonic wave vibrates the wire inside a magnetic field, which induces an electrical current. The ultrasound velocity map is then tomographically reconstructed by recording the current amplitude after translation and rotation of the wire. A hydrodynamic model provides a relationship between the velocity and the measured tension. Wire tension influence, electrical output characteristics, frequency response, sensitivity, directionality, and robustness to cavitation were characterized. A multi-wires hydrophone was also fabricated and tested. Results show that tension of the wire has negligible influence on the signal. No peak of electrical impedance was observed from 0.15 to 10 MHz. The signal was linear over pressure from 50 kPa to 15 MPa. The hydrophone was robust even when cavitation activity occurred. The directivity is explained with the Lorentz force expression. The multi-wire hydrophone could work only at low frequencies. Such hydrophone could be of interest for high intensity acoustic field characterization.

8:40

5aBAa3. Comparison of invasive and non-invasive methods of measuring high intensity acoustic fields. Claudio I. Zanelli, Samuel M. Howard, and Dushyanth Giridhar (Onda Corp., 1290 Hammerwood Dr., Sunnyvale, CA 94089, cz@ondacorp.com)

We present two methods of characterizing high intensity acoustic fields, namely, a non-invasive Schlieren method, and an invasive fiber-optic based one. The instant display makes the Schlieren method very attractive, although because it is a projection method it does not convey all the information available by the more localized sampling provided by the optical fiber. Numerical comparisons as well as the limitations of each method are described in the context of therapeutic ultrasound applications.

9:00

5aBAa4. Acoustic characterization and assessment of renal injury with a broad focal width electrohydraulic lithotripter. Yuri A. Pishchalnikov (Burst Labs., fka Impulse Devices, Inc., 13366H Grass Valley Ave., Grass Valley, CA 95945, yurapish@gmail.com), James A. McAteer, Bret A. Connors, Rajash K. Handa (Dept. of Anatomy and Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN), James E. Lingeman (Dept. of Urology, Indiana Univ. School of Medicine and Methodist Hospital Inst. for Kidney Stone Disease, Indianapolis, IN), and Andrew P. Evan (Dept. of Anatomy and Cell Biology, Indiana Univ. School of Medicine, Indianapolis, IN)

This study provides an independent assessment of a novel lithotripter (LG-380, Tissue Regeneration Technologies), marketed as having a long-life self-adjusting spark gap electrode and producing a low-pressure, broad-focal-zone acoustic field. For acoustic characterization we coupled the therapy head of the lithotripter to a water tank and mapped the field using a fiber-optic hydrophone (FOPH-500, RP Acoustics). At the target point of the lithotripter, the peak positive pressure (P+) remained relatively stable (~19 ± 5 MPa at power level 9) during the 6000 shock waves (SWs) lifetime of the electrode. The position of maximum P+ (~35 MPa at PL9) was 35 mm distal to target point and shifted progressively toward the therapy head as the electrode aged, reaching the target point (while reducing to P+~20 MPa) after ~5000 SWs. This was likely due to a slight movement in position of the self-adjusting spark gap—changing the focus of the shock wave and the dimensions of the focal volume of the lithotripter. Kidney injury was assessed using an established pig model by routine measures of renal function and quantitation of lesion size. Simulated clinical treatments (3000 SWs dose) damaged <0.1% functional renal volume, suggesting minimal potential for adverse effects with low-pressure broad-focal-zone lithotripters. [NIH DK43881.]

Contributed Papers

9:20

5aBAa5. Effect of *ex vivo* body wall on lithotripter shock waves. Guangyan Li, James A. McAteer, James C. Williams (Indiana Univ. School of Medicine, 635 Barnhill Dr., Indianapolis, IN 46202, gyli@iupui.edu)

Current understanding of mechanisms of shock wave (SW) action in shock wave lithotripsy comes almost entirely from laboratory testing. Past attempts to characterize lithotripter SWs *in vivo* have been hindered by difficulties in hydrophone alignment and physical constraints precluding precise mapping of the acoustic field. We adopted an *ex vivo* approach in which full-thickness segments of pig abdominal wall were secured against the inside surface of the Mylar acoustic window of a water-filled test tank coupled to a Dornier Compact-S lithotripter. A fiber-optic probe hydrophone was used to measure SW pressures and map the field in 2 mm steps in the focal plane. Peak positive pressure (P+) on axis was attenuated roughly proportional to tissue thickness (6.1% per cm, ~40% for a 5–6 cm body wall). Negative pressure (P−) was less affected. Shock rise time was unchanged (~18–20 ns). Although irregularities in tissue thickness affected SW focusing, step-wise mapping of the field showed no effect of the body wall on focal width (~6 mm with or without tissue). These findings suggest that apart from attenuation, the characteristics of the acoustic field for SWs passing through the body wall are remarkably similar to values collected in the free field. [NIH-DK43881.]

9:35

5aBAa6. Non-invasive estimation of temperature using diagnostic ultrasound during high intensity focused ultrasound therapy. Olga Bessonova and Volker Wilkens (Ultrasound Dept., Physikalisch-Technische Bundesanstalt, Bundesallee 100, Braunschweig 38116, Germany, olga.bessonova@ptb.de)

The use of HIFU for thermal ablation of human tissues requires safe real-time monitoring of the lesion formation during the treatment to avoid damage of the surrounding healthy tissues and to control temperature rise. Besides MR imaging, several methods have been proposed for temperature imaging using diagnostic ultrasound, and echoshift estimation (using speckle tracking) is the most promising technique. It is based on the thermal dependence of the ultrasound echo that accounts for two different physical phenomena: local change in speed of sound and thermal expansion of the propagating medium due to changes in temperature. In our experiments we have used two separate transducers: HIFU exposure was performed using a 1.06 MHz single element focusing transducer of 64 mm aperture and 63.2 mm focal length; the ultrasound diagnostic probe of 11 MHz operated in B-mode for image guidance. The temperature measurements were performed in an agar-based tissue-mimicking phantom complying with the specification given in IEC60601-2-5. To verify the obtained results, the numerical modeling of the acoustic and temperature fields was carried out using KZK and Pennes Bioheat equations. The comparison of the results simulated and measured, as well as measured with thermocouples, shows a rather good coincidence.

9:50

5aBAa7. Inertial cavitation enhancement using confocal ultrasound. Robert A. Fowler, Maxime Lafond, Adrien Poizat, Jean-Louis Mestas, Françoise Chavrier, Jean-Christophe Béra, and Cyril Lafon (Inserm U1032 LabTAU, Univ. of Lyon, Inserm U 1032, LabTAU, 151 Cours Albert Thomas, Lyon 69001, France, andrew.fowler@inserm.fr)

Inertial cavitation has been shown to be useful in many therapeutic applications; thus, controlling this phenomenon is of great therapeutic interest. However, the stochastic nature of cavitation often proves problematic for predicting its location and extent. Traditional solutions to this problem are the of use dedicated detection apparatuses, or to use injectable microbubbles (MBs), which act as nuclei for the initiation of cavitation. We hypothesize here that cavitation can be reliably controlled without the use of MBs using a confocal system, which produces a lobular focal zone due to acoustic interference. This interference pattern was studied both in simulation and hydrophone measurement. Cavitation extent was confirmed chemometrically with an assay for hydroxyl radical formation, and by passive cavitation detection with a hydrophone. A high speed camera was used to image the initiation of cavitation within the focal zone, the evolution of the bubble cloud, and the subsequent bubble rebound after pulse cessation. The experiments in this work confirm that cavitation is produced more reliably in the confocal setup as opposed to a single transducer, as well as illuminating the mechanisms for this enhancement. [Work supported by the European Union through the Eurostars program (project E!6173) and Caviskills SAS.]

10:05

5aBAa8. Scattering of high-intensity focused ultrasound by the ribs: Constrained optimization with a complex surface impedance boundary condition. Pierre Gelat (Acoust. and Ionising Radiation Div., National Physical Lab., Hampton Rd., Teddington TW11 0LW, United Kingdom, pierre.gelat@npl.co.uk), Gail ter Haar (Therapeutic Ultrasound Group, Joint Phys. Dept., Inst. of Cancer Res., Sutton, United Kingdom), and Nader Saffari (Dept. of Mech. Eng., Univ. College London, London, United Kingdom)

One of the challenges of trans-rib high-intensity focused ultrasound (HIFU) treatment is the need to transmit sufficient energy through the ribcage to ablate tissue whilst minimizing the formation of side lobes, and sparing healthy tissue. Ribs strongly absorb and reflect ultrasound. This may result in overheating of bone and overlying tissue during treatment, leading

to skin burns. Successful treatment of a patient with tumors of the liver or the pancreas therefore requires a thorough understanding of the way acoustic and thermal energy is deposited. A boundary element approach was developed to predict the field of a multi-element HIFU array scattered by human ribs in 3D. This forward model has been reformulated as a constrained least squares optimization problem where the velocity of each individual element on the array is an optimization variable. A locally reacting complex surface impedance condition at the surface of the ribs has been implemented. The methodology has been tested at an excitation frequency of 1 MHz on a spherical section multi-element array and a total of six array-rib configurations have been investigated. The results were compared against other focusing and rib-sparing approaches, demonstrating the efficacy and flexibility of the constrained optimization approach.

10:20–10:30 Break

10:30

5aBAa9. Validation of ultrasonic beam steering accuracy for medical applications by means of a modified Hough transform. Peter Ploß, Stefan J. Rupitsch, and Reinhard Lerch (Chair of Sensor Technol., Friedrich-Alexander-Univ. Erlangen-Nuremberg, Paul-Gordan-Str. 3/5, Erlangen 91052, Germany, peter.ploss@lse.eei.uni-erlangen.de)

Medical applications such as ultrasonography depend on the excitation of ultrasound wave packets that propagate with a targeted orientation in the examined body. Nowadays, the wavefronts' orientation is usually adjusted by the well-established electronic beam steering process, as compared to mechanical skewing of the ultrasound transducer. One sector scan (B-mode) is reconstructed from several A-mode acquisitions. This reconstruction process is dependent on the proper knowledge of the polar steering angles αk. From a clinical perspective, the correct reconstruction of the geometry is of great importance, since the detection of diseases (i.e., a tumor) is strongly dependent on the shape. We present a modified Hough transform that allows for analyzing acoustic pressure field distributions in order to precisely determine the direction of ultrasound propagation. The method is applicable to spatially discretized measurement and simulation data obtained either by hydrophone measurements or Finite Element Method simulations. Verification with artificial data confirmed an angular accuracy better 0.1°. This novel approach enables manufacturers of sonographic devices to verify the electronically chosen wavefront orientations by conducting measurements as well as simulations. The influence of parameters such as sampling rate, geometrical dimensions, material parameters, or apodization can be studied in a timesaving computer-aided engineering scheme.

10:45

5aBAa10. Intracardiac myocardial elastography in humans *in vivo* during radio-frequency ablation. Julien Grondin (Dept. of Biomedical Eng., Columbia Univ., 1210 Amsterdam Ave, ET 351, MC 8904, New York, NY 10027, ek2191@columbia.edu), Elaine Wan, Alok Gambhir (Dept. of Medicine - Div. of Cardiology, Columbia Univ., New York, NY), Stanley Okrasinski (Dept. of Biomedical Eng., Columbia Univ., New York, NY), Hasan Garan (Dept. of Medicine - Div. of Cardiology, Columbia Univ., New York, NY), and Elisa E. Konofagou (Dept. of Biomedical Eng., Columbia Univ., New York, NY)

Intracardiac echocardiography (ICE) is commonly used during radio-frequency (RF) ablation procedures for procedural guidance. Besides its imaging function, ICE could be used to assess mechanical properties of the myocardium to improve the ablation outcome. The objective of this study was to demonstrate the feasibility of imaging myocardial strains *in vivo* within the same imaging plane as ICE at high temporal resolution. A 5.8-MHz center frequency ICE probe was used to image the heart of two humans with atrial arrhythmias *in vivo* before and after RF ablation at high frame rates (1200 Hz), and the channel data were acquired on a clinical ultrasound system. The RF signals were reconstructed on a 9cm depth and 90° field of view region and axial cumulative displacement estimation was performed using 1-D cross-correlation using a window size of 2.6 mm and 95% overlap. Cumulative axial strains were obtained from the displacements using a least-squares estimator with a kernel of 5.1 mm. Cumulative axial strains in the left atrium during systole were 23% and 18% in the two subjects before ablation, changing to 8% and 11% in the same location after ablation. Myocardial elastography could thus provide some quantitative methods for monitoring the generation of thermal lesions.

11:00

5aBAa11. Simulation of diagnostic ultrasound imaging with the fast nearfield method. Yi Zhu, Peter B. Beard, and Robert J. McGough (Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48864, mcgough@egr.msu.edu)

Software for simulating diagnostic ultrasound imaging with the fast nearfield method is now available in FOCUS (http://www.egr.msu.edu/~fultras-web). The implementation of ultrasound imaging simulations in FOCUS first decomposes the excitation pulse into two shorter pulses that, when convolved together, are equivalent to the original excitation pulse. These two shorter pulses are applied to the transmit and receive apertures, reciprocity is applied to the receive aperture, and then the results of the intermediate pressure calculations are convolved to obtain the simulated pulse-echo signal. Simulation results using point scatterers are compared to Field II (http://field-ii.dk/) for simulations with a linear phased array. Results show that FOCUS achieves equal or smaller errors in less time than Field II, and the FOCUS results require much smaller sampling frequencies. The FOCUS results are achieved with an analytically equivalent algorithm that avoids aliasing problems by calculating pressures with the fast nearfield method combined with time-space decomposition. To further improve the efficiency of the imaging simulation in FOCUS, intermediate signals are computed, stored, and re-used. Ongoing efforts to incorporate new features into ultrasound imaging simulations with FOCUS will also be discussed. [This work was supported in part by NIH Grant R01 EB012079.]

11:15

5aBAa12. Progress in inferring desmoplastic stromal tissue microstructure noninvasively with ultrasound nonlinear elasticity imaging. Elizabete R. Ferreira (Civil Eng., Univ. of Minnesota, Minneapolis, MN), Assad A. Oberai (Mech., Aerosp., and Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY), Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St, Boston, MA 02215, barbone@bu.edu), and Timothy J. Hall (Medical Phys., Univ. of Wisconsin, Madison, WI)

Desmoplasia is the abnormal growth of fibrous tissue that is often associated with malignancy in breast and other solid tumors. The collagen comprising the extracellular tissue matrix tends to be more dense than in normal breast tissue, and to exhibit altered microstructure. The distinctive mechanical properties of solid tumors are often attributed to the abnormal density and microstructure of their collagen extracellular matrices. For example, a higher concentration of collagen implies a larger linear elastic modulus, while a decrease in fiber tortuosity would mean a smaller "toe" region in the stress-strain curve. This talk describes progress of inferring local tissue microstructural properties from noninvasive, *in vivo* measurement of nonlinear elastic properties of breast tissue. An ultrasound elasticity-imaging based approach for determining the average local micro-structural properties of tissue was developed and implemented. It utilizes multiscale homogenization methods to determine effective macroscopic properties from assumed microstructural arrangements of collagen fibers. The resulting nonlinear constitutive equation is then used in a model-based elastic inverse problem solver to obtain local microstructural tissue properties from macroscopic observations of tissue deformation. Its potential in diagnosing malignant breast tumors in a small set of patients was also examined. [NSF Grant 50201109; NIH NCI-R01CA140271.]

11:30

5aBAa13. Analysis and measurement of the modulation transfer function of harmonic shear wave induced phase encoding imaging. Stephen A. McAleavey (Biomedical Eng., Univ. of Rochester, 309 Goergen BME/Optics Bldg., Rochester, NY 14627, stephen.mcaleavey@rochester.edu)

B-scan imaging relies on geometric focusing of ultrasound echoes. In contrast, the method of Shear Wave Induced Phase Encoding imaging achieves lateral resolution of an elastic target by using traveling shear waves to encode the position of scatters in the phase of the received echo. A Fourier series description of the phase modulated echo signal is developed, demonstrating that echo harmonics at multiples of the shear wave frequency reveal target k-space data at identical multiples of the shear wavenumber. Modulation transfer functions (MTFs) are calculated for maximum shear wave acceleration and maximum shear constraints, and compared with a conventionally focused aperture. The relative SNR of this method versus a conventionally focused aperture is determined from these MTFs. Reconstructions of wire targets in a gelatin phantom using a cylindrical shear wave source are presented, including Images generated from the fundamental and second harmonic of the shear wave modulation frequency. Comparison of images generated at 1 and 3.5 MHz demonstrate the weak dependence of lateral resolution on ultrasound frequency with this method.

11:45

5aBAa14. Estimation of nonlinear parameters applying uniaxial and shear stress to inhomogeneous viscoelastic media. Timofey Krit (Phys., Moscow State Univ., Leninskie Gory, Bldg. 1/2, Moscow, Moscow 119991, Russian Federation, timofey@acs366.phys.msu.ru)

Static shear deformations of a plane-parallel layers of several viscoelastic media created simultaneously with the uniaxial compression are considered. Each layer is fixed between two rigid plates. Displacement of one plate relative to the other resulted in shear strain of the layer. This strain could reach 0.6 of the layer thickness. At such strain, effects due to the cubic nonlinearity arise. It is shown that measuring the dependence of the shear stress on the shear strain along one axis at different compression along the perpendicular axis one could determine nonlinear Landau parameters. The measurements were performed in layers of plastisol, gelatin, and farina-gelatin of 7 mm thickness with a rectangular base 8.9×8.9 cm, mounted between three aluminum plates. The upper plate was loaded with masses ranging from 0 to 25 kg and was fixed in each series of the stress-strain measurements. The values of the Landau coefficient A were measured in layers with different value of linear shear modulus. The different behavior of stress-strain curves was observed for media with different composition. [Work supported by the Russian Foundation for Basic Research (Grant Nos. 12-02-00114 and 12-02-31418), and grant of the Government of the Russian Federation 11.G34.31.0066.]

FRIDAY MORNING, 6 DECEMBER 2013 GOLDEN GATE 4/5, 8:00 A.M. TO 10:00 A.M.

Session 5aBAb

Biomedical Acoustics: High Frequency Ultrasound

Jeffrey A. Ketterling, Chair
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Contributed Papers

8:00

5aBAb1. Cyclic variation of three-dimensional geometry of the rat carotid artery bifurcation assessed by high-frequency ultrasound imaging. Changzhu Jin, Kweon-Ho Nam, Tae-Hoon Bok, and Dong-Guk Paeng (Ocean System Eng., Jeju National Univ., Rm. 5464 of No.4 Bldg., Ocean Sci. College, Jeju City 690-756, South Korea, yustchang@jejunu.ac.kr)

The computational simulation of the interaction between blood flow and arterial wall motion during a cardiac cycle is complicated and requires high quality information of the vessel wall motion in space and time. In this study, a set of cross-sectional ultrasound images was acquired using an electrocardiogram-gated kilohertz visualization mode, which provides 1000 frame images per second, by an ultrasound imaging system (Vevo 770, VisualSonics, Canada) with a probe of 40-MHz central frequency (RMV 704). The three dimensional (3D) geometry of carotid artery bifurcation was reconstructed at systolic and diastolic phases during a cardiac cycle using the cross-sectional ultrasound images, and a block meshing method was applied to the reconstructed 3D geometry. Then, an appropriate hexahedral mesh was constructed for the computational simulation. The 3D geometry measured by high-frequency ultrasound imaging provides high temporal and spatial information on the vessel wall motion during a cardiac cycle, which is important for accurate computational simulation of hemodynamics in the bifurcation area of the carotid artery. The fluid-structure interaction simulation of blood flow in the carotid artery bifurcation during a cardiac cycle is in progress using the meshes of 3D geometry of vessel wall. [Work supported by NIPA-2013-H0401-13-1007 and 2013R1A1A2043478.]

8:15

5aBAb2. High-frequency ultrasound of breast tissue phantoms with histology mimicking microstructures. Audrey P. Butler, Robyn K. Omer (Biology, Utah Valley Univ., 800 W. University Parkway, M.S. 299, Orem, UT 84058, audreyphoenix@gmail.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The objectives of this study were to develop breast tissue phantoms with microstructures that accurately mimic the histology of normal and malignant tissue, and to determine the effects of these microstructures on high-frequency (HF) ultrasonic spectra (10–100 MHz). Phantoms were created from a mixture of water, gelatin, and soluble fiber. To simulate various breast tissue histologies, polyethylene beads and nylon fibers with a range of diameters were embedded into phantoms. Microstructures ranging from simply dispersed beads to bead-fiber constructs resembling terminal ductal lobular units were modeled and tested. Pitch-catch and pulse-echo measurements were acquired using 50-MHz transducers, a HF pulser-receiver, and a 1-GHz digital oscilloscope. Spectra were derived from the data and peak densities (the number of peaks and valleys in a specified spectral range) were determined from the spectra. Preliminary results from dispersed beads (58–925 μm diameter) of constant volume concentration (0.8%) indicated that the smaller beads produced higher peak densities than the larger beads with a consistent trend. The higher peak densities can be attributed to either the higher number of scatterers for small beads or the size of scatterer in relation to the ultrasonic wavelength. These and results from more advanced histologically accurate microstructures will be discussed.

8:30

5aBAb3. Molecular subtyping of breast cancer *in vitro* using high-frequency ultrasound. Janeese E. Stiles (Biology, Utah Valley Univ., 800 W. University Parkway, M.S. 299, Orem, UT 84058, stilesjaneese@gmail.com), Laurel A. Thompson (Chemistry, Utah Valley Univ., Orem, UT), Mandy H. Marvel, Janice E. Sugiyama (Biology, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The molecular subtypes of breast cancer correlate more strongly to prognosis, treatment response, and local recurrence than the traditional classifications based on histopathology. The ability to determine the subtype of breast tumors during surgery or biopsy in real time would provide physicians with new diagnostic capabilities to screen suspicious lesions, to perform high-precision surgery on malignant and premalignant lesions, and to personalize treatment for patients. This work studied the potential of using the molecular subtypes as natural biomarkers for characterizing breast cancer with high-frequency ultrasound (10–100 MHz). We hypothesized that high-frequency ultrasound would be able to detect variations in cell biomechanical properties due to mutations found in aggressive subtypes (e.g., basal-like and Her2 +). These mutations alter the expression levels of proteins that regulate the actin cytoskeleton, thereby modifying the biomechanical and thus ultrasonic properties of the cells. Pulse-echo measurements were acquired *in vitro* from breast cancer cell lines with different subtypes. The results showed that each cell line produced a unique ultrasonic spectral signature. One of the cell lines additionally exhibited changes over time, possibly due to dedifferentiation. Correlation of the results to other cell characterization methods will also be presented. [This work was supported by Utah Valley University.]

8:45

5aBAb4. Characterizing breast cancer cell lines using principal component analysis of high-frequency ultrasonic spectra. Laurel A. Thompson (Chemistry, Utah Valley Univ., 800 W. University Parkway, M.S. 179, Orem, UT 84058, laurelathompson@gmail.com), Janeese E. Stiles, Mandy H. Marvel, Janice E. Sugiyama (Biology, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Current methods for determining the molecular subtypes of breast cancer include DNA microarrays, immunohistochemical staining, and proteomic analysis. These methods are not easily transferable to real-time clinical applications such as the intraoperative assessment of margin or biopsy specimens. The development of a diagnostic method for rapidly determining the molecular subtype of malignant breast tissue in a clinical setting would represent a significant advance in breast cancer detection and treatment. Preliminary results suggest that high-frequency ultrasound may be sensitive to variations between breast cancer cell lines, and thus molecular subtype, due to a mechanism linking subtype mutations to the ultrasonic properties of the cells. This mechanism was explored using an integrated experimental and computational method. Ultrasonic scattering from breast cancer cells *in vitro* were simulated using a multipole-based approach. Variations between cell lines due to different molecular subtypes were modeled using a range of cell moduli and sizes. Model and experimental spectra were compared using principal component analysis (PCA). The results indicate the properties and thus molecular subtypes of breast cancer cells could potentially be determined by comparing their measured spectra to model spectra using a feature classification program such as PCA. [This work was supported by Utah Valley University.]

9:00

5aBAb5. A quantitative assay of neovascularization using high-frequency ultrasonic spectroscopy. Michaelle A. Cadet, Andrea N. Quiroz, Janeese E. Stiles (Biology, Utah Valley Univ., 800 W. University Parkway, M.S. 299, Orem, UT 84058, michaelle.alexandra@gmail.com), Laurel A. Thompson (Chemistry, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The stimulation and inhibition of tissue vascularization has important applications to tissue engineering and oncology. Approaches to quantitatively evaluate neovascularization *in vivo* in adult animals with differentiated tissue include both invasive methods that use an implanted or injury-induced matrix in the study organism, or noninvasive small animal imaging methods such as MRI, CT, and PET. The objective of this study was to determine if ultrasonic spectra in the 10–100 MHz range could be used as an *in vivo* neovascularization assay. Numerical simulations and phantoms were used as model systems to test the feasibility of the approach. The simulations modeled ultrasonic scattering from microscopic vascular networks using clusters of walled, randomly oriented cylinders to represent blood vessels and cylindrical wave functions to represent ultrasonic waves. Phantoms with microscopic channels were fabricated from gelatin, soluble fiber, and thin polymer strands embedded into the gel. The strands were removed after the gel solidified and filled with ultrasonic blood simulant. Initial numerical results indicated that increasing the number of blood vessels in tissue significantly altered the spectra. Methods for analyzing the data, including the use of spectral parameters and pattern recognition programs such as principal component analysis, will be discussed.

9:15

5aBAb6. Determining affects of temperature change on tissue using high-frequency ultrasound: Porcine heart (aorta). David S. Stroud (Phys., Utah Valley Univ., 581 West 480 North, American Fork, UT 84003, stroud.david@yahoo.com), Chad Haskel, and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The purpose of this study was to determine if high-frequency (HF) ultrasound (20–80 MHz) is sensitive to the change of average human body in temperature (approximately 37 °C) to average room temperatures of approximately 20 °C to 25 °C. When temperatures decrease, water molecules within organic tissue expand, decreasing their density. Changes in temperatures can alter the number of molecules found within a given area, altering tissue density. Two parameters in high-frequency ultrasound (20–80 MHz) have been found to be sensitive to a range of pathologies in resected margins from breast conservation surgery: The number of peaks (the peak density) in the waveform spectrum and the slope of the Fourier transform of the waveform spectrum. Changes in temperatures may affect the accuracy of these two parameters. To test this hypothesis, through-transmission and pulse-echo measurements were acquired fresh (within one hour of slaughter from local butchers) aorta samples from porcine hearts. Results will be presented and discussed. [This work was supported by a Utah Valley University Presidential Fellowship Award.]

9:30

5aBAb7. High-frequency ultrasound study of excised tissue cryopreserved via simple sugars. Christopher D. Sutherland, Logan L. Warner (Biology, Utah Valley Univ., 800 W. University Parkway, M.S. 299, Orem, UT 84058, sutekinachris@hotmail.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

High-frequency ultrasound (20–80 MHz) has been found to be sensitive to a range of pathologies in excised breast tissue before fixation in formalin or other formaldehyde analogues. Formalin fixation, however, may alter the structure and rigidity of a sample so that data gathered using high-frequency ultrasound after fixation may no longer be viable for research purposes. This limits the amount of time researchers may conduct tests, so preservation via simple sugars is being considered. Numerous studies have been conducted using sucrose, trehalose, or glucose as cryoprotectants for cells and simple tissues. The objective of this study was to test the sensitivity of high-frequency ultrasound to changes in the microstructure, stiffness, and cellular integrity of tissue samples due to cryopreservation with these sugars. Domestic pig heart tissue was placed in aqueous solutions of sucrose, trehalose, and D-(+)-glucose. The specimens were refrigerated and observed over time using high-frequency ultrasound to detect tissue damage. The results of this study suggest that cryopreservation with sugars will not only allow more time for researchers to conduct ultrasonic tests on surgical specimens, but also that high-frequency ultrasound could potentially be used as an assay to measure tissue degradation in preserved living tissues such as transplant organs.

9:45

5aBAb8. Correlation between nano- and macro-properties of cell walls. Bernhard R. Tittmann and Xiaoning Xi (Eng. Sci. & Mech., Penn State Univ., 212 EES Bldg., University Park, PA 16802, brt4@psu.edu)

This report is on the imaging and characterization of the nano-composite structure of native primary cell walls. In particular, the structure of primary celery epidermis cell walls was imaged with sub-nanometer resolution using the atomic force microscope (AFM). The high-frequency acoustic microscope was used to image onion cell (*Allium cepa*) wall epidermis at the micro-scale at 600 MHz at 1 micron resolution. V(z) signatures were obtained and used to estimate the bulk modulus of 3.30 GPa in good agreement with destructive measurements. The combined results reveal a surprisingly fine but strong nano-composite for the lignocellulosic biomass.

FRIDAY MORNING, 6 DECEMBER 2013 MASON, 8:30 A.M. TO 11:00 A.M.

Session 5aEA

Engineering Acoustics: General Engineering Acoustics: Mufflers, Waveguides, Materials, and Other Topics

Roger T. Richards, Chair
US Navy, 169 Payer Ln., Mystic, CT 06355

Contributed Papers

8:30

5aEA1. Mutual validation of muffler performance evaluation methods using field measurements and finite element analysis. Scott Mang and Jeffrey S. Vipperman (Mech. Eng., Univ. of Pittsburgh, 3700 O'Hara St, Pittsburgh, PA 15213, jsv@pitt.edu)

The design of large industrial mufflers poses a problem to many companies involved in their fabrication. Most of the design and evaluation theory has been developed using one dimensional acoustics. However, it is well known that in large ducts, the one dimensional propagation assumption breaks down at much lower frequencies. The three dimensional propagation of acoustic waves in complex geometries is a very difficult problem to solve analytically. For this reason, numerical methods are employed. In this study, a finite element analysis (FEA) program is validated using field measurements. Three well known muffler performance criteria and three different large sized mufflers are considered. The three performance metrics are noise reduction (NR), insertion loss (IL), and transmission loss (TL). During the study, sufficient agreement for validation of the performance evaluation methods was found. A method for reconciling idealistic FEA simulations and *in situ* measurements by accounting for significantly high ambient noise is developed.

8:45

5aEA2. Study of micro-perforated tube mufflers with adjustable transmission loss. Longyang Xiang, Shuguang Zuo, Menghao Zhang, Jiajie Hu, and Guo Long (Clean Energy Automotive Eng. Ctr., Tongji Univ., No. 4800, Cao'an Hwy., Shanghai, China, Shanghai 201804, China, longyang.x@gmail.com)

Micro-perforated tube muffler has the advantages of both the reactive and dissipative mufflers. The paper proposed that the change of micro-perforated tube length could change the transmission loss behaviors of the micro-perforated tube mufflers. The acoustic impedance of micro-perforated plate was derived in this paper, which was used in the finite element method to calculate the transmission loss of mufflers with various micro-perforated tube lengths. Then, a relation formula of the maximum muffling frequency and the micro-perforated tube length was fitted. This kind of adjustable mufflers could be used efficiently to control the noise of the rotating machines, whose main noises are always proportional to the rotating speed.

9:00

5aEA3. Evaluation of a collocation method for the analysis of multi-segmented waveguides at high-frequencies. Jerry H. Ginsberg (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338-2854, jerry.ginsberg@me.gatech.edu)

A method for analyzing high-frequency signal propagation in rectilinear waveguides consisting of interconnected segments has been proposed [Ginsberg, POMA **19**, 030104 (2013)]. The signal within each segment is represented as a series of modes that either propagate or evanesce in the forward and backward directions. The state variables are the modal amplitudes. Equations for these parameters are obtained by satisfying continuity and boundary conditions on pressure and axial particle velocity at preselected points on transitional cross-sections. This leads to a sequential evaluation of transfer functions for the modal amplitudes. The present work will examine the convergence and accuracy properties of the formulation by comparing its results to those obtained from a classical orthogonality analysis. The system consists of a pair of coaxial circular hard-walled cylinders axisymmetrically excited by a piston covering half the radius at one end, and the other end is rigidly closed. The frequency for the evaluation is slightly greater than the cutoff value for the sixth mode in the narrower segment. Aspects that are examined are computer code verification, metrics for comparing analyses, the influence of the number and placement of the collocation points, and the influence of evanescent modes.

9:15

5aEA4. Validation of two-port temperature models for heat dissipation in exhaust systems. Weam S. Elsahar, Tamer M. Elnady (Group for Adv. Res. in Dynamic Systems (ASU-GARDS), Faculty of Eng., Ain Shams Univ., 1 Elsarayat St., Abbaseya,Cairo 11517, Egypt, weam@eng.asu.edu.eg), and Mats Åbom (Marcus Wallenberg Lab. for Sound and Vib. Res., The Royal Inst. of Technol. (KTH), Stockholm, Sweden)

The acoustic performance of exhaust systems is affected by the flow speed and temperature of the exhaust gases. The flow speed affects the convective wave speed and changes the acoustic properties of some acoustic elements such as perforates. The temperature of the exhaust gas affects its density and the speed of sound. It is important to model the flow and temperature distribution within the exhaust system in order to perform an accurate acoustic simulation. The acoustic propagation along an exhaust system can be modeled by dividing the exhaust system into a number of two-port elements. It has been previously shown that flow distribution and pressure drop calculation can be done using the same two-port elements but using a different set of transfer matrices. A similar approach was previously proposed to calculate the temperature drop using the same two-port elements but using a new set of transfer matrices. This technique has the advantage to solve all physics by defining only one network. An improved version of this formulation is presented in this paper. Several cases were tested and the thermal and acoustic results were compared to finite element simulations and measurements. The two port results matched well with the other techniques.

9:30

5aEA5. Modeling acoustic propagation in a compartment fire. Mustafa Z. Abbasi, Preston S. Wilson, and Ofodike A. Ezekoye (Appl. Res Lab. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E Dean Keeton st., Austin, TX 78751, mustafa_abbasi@utexas.edu)

Firefighters unable to move and in need of rescue use an audible alarm to signal for help. Rescue teams can then follow this sound to the firefighter. This alarm is governed by NFPA 1982 : Standard on Personal Alert Safety System (PASS). Introduced in 1983, the PASS has saved many firefighter lives. However, a number of incidents have occurred where the PASS is less effective. There have been incidents where the PASS was heard sporadically on the fireground, or where localization of the alarm was difficult, leading to injury and loss of life. We hypothesized that the temperature field created by the fire is distorting the sound, making it difficult to recognize and localize. At ICA 2013, the authors presented experimental results showing changes in the room acoustic transfer function as the fire evolved. This paper will present efforts at modeling these effects. Using a combination of computational fluid dynamics and wave models, a comprehensive model will be presented capable of modeling sound propagation in the firefighting environment. The goal of this work is to develop a PASS signal more robust against distortion by the fire, and better able to serve the firefighting community. [Work supported by DHS/FEMA.]

9:45

5aEA6. A wearable real-time vocal biofeedback device. Mark VanDam (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu), Bradley C. Clemetson, Marshall Hurson, Tyler Pacheco (School of Eng. and Appl. Sci., Gonzaga Univ., Spokane, WA), Shirley Jakubowski, Walter Jakubowski (Parkinson's Resource Ctr. of Spokane, Spokane, WA), and Doreen Nicholas (Commun. Disord., Eastern Washington Univ., Spokane, WA)

A body-worn, real-time speech and voice biofeedback device is described. Data from an acoustic microphone and piezoelectric sensor worn comfortably in a neckband are streamed to a digital signal processor and a small, mobile computer, altogether able to fit into a pocket for extended use. User laryngeal and spectral characteristics are determined from the combination of sensor inputs. Selected vocal characteristics (e.g., vocal intensity, shimmer, jitter, spectral output, and fundamental frequency) are analyzed in real-time to provide immediate user feedback via tactile or visual response to indicate speech production pathologies including reduced loudness, pitch instability, or other features. With minimal training, this feedback can be immediately acted upon by the wearer to adjust speech and voice production characteristics accordingly. In addition, all data from input sensors are collected and stored in the computer's memory for offline analyses of speech and voice production characteristics. Extended-use, large-sample data collection addresses issues in the extant literature including ecological validity, reactivity (e.g., the Hawthorne effect), small sample sizes, and unaccounted for individual differences. This work offers a realistic description of voice use and assesses a wide range of functional and organic clinical conditions that are known to affect speech production (e.g., Parkinson's disease).

10:00–10:15 Break

10:15

5aEA7. Measurements of sound absorption of living grass. Chelsea E. Good, Aldo J. Glean, Joseph F. Vignola, John A. Judge, Teresa J. Ryan, Nicole B. Bull (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, 26good@cardinalmail.cua.edu), and Diego Turo (BioEng., George Mason Univ., Fairfax, VA)

This work presents measurements of acoustic absorption coefficients of sod samples with grass blades of different length. These measurements were made with a vertical acoustic impedance tube over a 200–1600 Hz frequency band. The acoustic measurements will be compared to values calculated using an equivalent fluid model. Sod is a coarse aggregate material such that the observed absorption will have components resulting from the different constituent elements. A layer of granular material with known acoustic properties is placed at the bottom of each sample to account for the acoustic absorption of soil. This work considers the sod as a two-component system: foliage and substrate (soil and roots together). The absorption effects due to each component were isolated by making measurements before and after shearing the mature foliage near the soil surface. We show the effects of foliage length on acoustic absorption. Measurements of sound absorption of living grass

10:30

5aEA8. Acoustic characteristics of a dielectric elastomer absorber. Zhenbo Lu, Yongdong Cui (Temasek Labs., National Univ. of Singapore, T-Lab Bldg., 5A, Eng. Dr. 1, #09-02, Singapore 117411, Singapore, tslluz@nus.edu.sg), Jian Zhu, Zijie Zhao (Dept. of Mech. Eng., National Univ. of Singapore, Singapore, Singapore), and Marco Debiasi (Temasek Labs., National Univ. of Singapore, Singapore, Singapore)

The present paper is devoted to study the acoustic characteristics of a dielectric elastomer (DE) absorber, which has a wide variety of potential applications as a novel actuator technology. DE, a lightweight and high elastic energy density smart material, can produce a large deformation under high DC/AC voltages. These excellent characteristics can be used to improve the present typical noise control systems. The performance of using this new soft-controlled-material is experimentally investigated. It is found that the voltage on the DE could tune the resonance frequencies of DE absorber thus it could absorb broadband noise. The results also provide insight into the appropriateness of the absorber for possible use as an active noise control system for replacing the traditional acoustic treatment.

10:45

5aEA9. Underwater acoustic measurements of an ultrasonic barrier for guidance of American Shad in front of hydroelectric installations. Francois Lafleur (EMMH, Hydro-PQ Res. Inst., 1800 boul. Lionel-Boulet, Varennes, QC J3X1S1, Canada, lafleur.francois@ireq.ca)

This article presents the results of ultrasonic underwater acoustic measurements in a project guide shads during their downstream migration. Biological issues will be explained to allow the context, but the focus will be on the acoustic problems. In the spring, thousands of shad ascend the St. Lawrence River to spawn downstream of the Central Carillon. After spawning, adults return to sea heading toward the dam Rivière-des-Prairies. The configurations of the installed barriers at the Rivière-des-Prairies dam and at the ile Bizard site will be presented. A design of a signal amplifier was performed to optimize the barrier. A series of simulations and acoustic measurements have been conducted for the evaluation of the emission level of the barrier. The measurement strategy must take into account aspects such as high frequency signal (125 kHz) and geolocation measurement to allow achieving a mapping program of the barrier. The paper will describe: The issue biological; The deployment sites; The mechanism of hearing ultrasonic shad; Configuration of the barrier; The measurement system and the analysis of the results; Typical results obtained for mapping acoustic; Future directions in terms of signal measurements of the acoustic barrier.

Session 5aMU

Musical Acoustics and Structural Acoustics and Vibration: Computational Methods in Musical Acoustics I

Edgar J. Berdahl, Chair
Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803

Invited Papers

10:00

5aMU1. Computational simulation of clarinet-like models. William J. Strong (Phys. & Astronomy, Brigham Young Univ., C126 ESC, Provo, UT 84604, strongw@byu.edu)

The presentation will review the computational simulation of two clarinet-like models. The Stewart and Strong model (J. Acoust. Soc. Am. **68**, July 1980) consisted of a uniform cylindrical tube with an attached clarinet mouthpiece and reed. The tube was represented by 0.25-cm sections of lumped element transmission line. The tapered mouthpiece was similarly represented with 0.1-cm sections. The reed was represented as a non-uniform bar clamped at one end. The simulation was carried out on a DEC PDP-15 computer and had a running time of roughly 250,000 times real time. The Sommerfeldt and Strong model (J. Acoust. Soc. Am. **83**, May 1988) was similar to the Stewart model, but included seven toneholes. The impedance of the tube with mouthpiece was calculated and used to calculate the impulse response. This was used to determine the interaction between pressure and flow at the reed opening. A player's airway was also included in the model because, at the time, there was some interest in its effect on tone production. Computational and experimental results will be presented. The two examples will illustrate some limits to simulation of models when computational power is inadequate as was true at the time.

10:20

5aMU2. Damping in turbulent Navier-Stokes finite element model simulations of wind instruments. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de)

The flute has a very inefficient energy use, where $<1\%$ of the blowing energy ends in the produced sound caused by the high impedance of the blowing hole suggesting a heavy damping in the system. Such damping is known from turbulent flows with high Reynolds numbers. In a FEM simulation of the Navier-Stokes equation of the flute, the inflow of blowing velocity into the flue shows unrealistic values. An alternative is the reasoning of Kolmogorov about turbulent flow as a cascade of vortices satisfying the scaling law as a steady-slope in a log-log relation of vortices size and damping. This is incorporated in the Navier-Stokes model as Reynolds-Averaged-Navier-Stokes model. In a FEM simulation of the flute using this model, the inflow of energy into the flute is modeled realistically, pointing to damping as the crucial factor in flute sound production. As a second example, a transient FEM model of a flow-structure coupling of a saxophone mouthpiece also shows strong damping of the inflow into the mouthpiece due to strong turbulence in the mouthpiece. Only because of this strong damping the system coupled to a tube results in stable oscillations and therefore in a stable tone production.

10:40

5aMU3. Bowed string simulation combining digital waveguide and finite-difference time-domain frameworks. Esteban Maestre (Information and Commun. Technologies, Universitat Pompeu Fabra, Roc Boronat 138, Barcelona 08018, Spain, esteban.maestre@upf.edu), Carlos Spa (Math, Universidad Federico Santa Maria, Santiago, Chile), and Julius Smith (Music, Stanford Univ., Stanford, CA)

In light of the promising results obtained by driving a low-complexity digital waveguide (DW) violin model with synthetic bowing gestures, we are currently exploring the possibilities of combining DW and finite-difference time-domain (FDTD) frameworks to construct refined physical models of string quartet instruments. Departing from state-of-the-art bowed string simulation paradigms, we extend previous approaches by combining a finite-width bow-string interaction model and a dynamic friction model based on simulating heat diffusion along the width of the bow. In our model, DW is used for string propagation, while FDTD is used for bow-string interaction and heat diffusion. The bridge termination is realized using an efficient, passive digital filter obtained from admittance measurements. In this talk, we will present and discuss the current status and future directions of our modeling work, including two-dimensional string vibration simulation for horizontal and vertical polarizations.

11:00

5aMU4. A Hamiltonian approach to simulation of acoustic systems involving nonlinear interactions. Vasileios Chatziioannou (Inst. of Music Acoust., Univ. of Music and performing Arts Vienna, Anton-von-Webern-Platz 1, Bldg. M, Vienna 1030, Austria, chatziioannou@mdw.ac.at) and Maarten van Walstijn (Sonic Arts Res. Ctr., Queen's Univ. Belfast, Belfast, United Kingdom)

Nonlinear interactions take place in most systems that arise in music acoustics, usually as a result of player-instrument coupling. Several time-stepping methods exist for the numerical simulation of such systems. These methods generally involve the discretization of the Newtonian description of the system. However, it is not always possible to prove the stability of the resulting algorithms, especially

when dealing with systems where the underlying force is a non-analytic function of the phase space variables. On the other hand, if the discretization is carried out on the Hamiltonian description of the system, it is possible to prove the stability of the derived numerical schemes. This Hamiltonian approach is applied to a series of test models of single or multiple nonlinear collisions and the energetic properties of the derived schemes are discussed. After establishing that the schemes respect the principle of conservation of energy, a nonlinear single-reed model is formulated and coupled to a digital bore, in order to synthesize clarinet-like sounds.

11:20

5aMU5. A modal architecture for artificial reverberation. Jonathan S. Abel, Sean Coffin (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., 660 Lomita Dr., Knoll 306, Stanford, CA 94305, seancoff@ccrma.stanford.edu), and Kyle S. Spratt (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

The method of analyzing an acoustic space by way of modal decomposition is well established. In this work, a computational structure employing modal decomposition is introduced for synthesizing artificial reverberation, implementing the modes using a collection of resonant filters, each driven by the source signal and summed in a parallel structure. With filter resonance frequencies and dampings tuned to the modal frequencies and decay times of the space, and filter gains set according to the source and listener positions, any number of acoustic spaces and resonant objects may be simulated. While convolutional reverberators provide accurate models but are inflexible and computationally expensive, and delay network structures provide only approximate models but are interactive and computationally efficient, the modal structure presented in this work provides explicit, interactive control over the parameters of each mode, allowing accurate modeling of acoustic spaces, movement within them and morphing among them. Issues of sufficient modal density, computational efficiency and memory use are discussed. Finally, models of measured and analytically derived reverberant systems are presented, including a medium-sized acoustic room and an electro-mechanical spring reverberator.

11:40

5aMU6. Real-time finite difference-based sound synthesis using graphics processors. Marc Sosnick-Perez and William Hsu (Comput. Sci., San Francisco State Univ., 1600 Holloway Ave., San Francisco, CA 94114, whsu@sfsu.edu)

Finite difference methods can be used to model the vibrations of plates and membranes; the output of these numerical simulations can then be used to generate realistic and dynamic sounds. To create interactive, playable software instruments with finite difference methods, we need to be able run large simulations in real-time. Real-time finite difference-based simulations of large models are typically too compute-intensive to run on CPUs. The ubiquity of graphics processors (GPUs) today make them obvious choices for speeding up such applications. We have implemented finite difference simulations that run in real-time on GPUs. We will describe how we address the problems that arise from interactions between real-time audio constraints and GPU architecture and performance characteristics, and demonstrate the current version of FDS, our Finite Difference Synthesizer.

FRIDAY MORNING, 6 DECEMBER 2013 — CONTINENTAL 9, 8:00 A.M. TO 11:15 A.M.

Session 5aNS

Noise: Perception and Measurement of Sound and Sound and Equipment

Steven D. Pettyjohn, Chair
The Acoust. & Vib. Group, Inc., 5700 Broadway, Sacramento, CA 95820

Contributed Papers

8:00

5aNS1. Laboratory study of outdoor and indoor annoyance caused by sonic booms from sub-scale aircraft. Alexandra Loubeau, Jonathan Rathsam, and Jacob Klos (Structural Acoust. Branch, NASA Langley Res. Ctr., M.S. 463, Hampton, VA 23681, a.loubeau@nasa.gov)

Advances in integrated system design tools and technologies have enabled the development of supersonic aircraft concepts that are predicted to produce sonic booms with lower loudness levels while maintaining aerodynamic performance. Interest in the development of a low-boom flight demonstration vehicle for validating design and prediction tools and for conducting community response studies has led to the concept of a sub-scale test aircraft. Due to the smaller size and weight of the sub-scale vehicle, the resulting sonic boom is expected to contain spectral characteristics that differ from that of a full-scale vehicle. In order to justify the use of sub-scale aircraft for community annoyance studies, it is necessary to verify that these spectral differences do not significantly affect human response. The goal of the current study is to evaluate both outdoor and indoor annoyance caused by sonic booms predicted for these two classes of vehicles. The laboratory study is conducted in two sonic boom simulators that provide a realistic reproduction of the outdoor and indoor sonic booms. The indoor facility also provides a realistic listening environment to address the effect on human annoyance of interior rattle noises predicted to be induced by the structural excitation from the sonic boom.

8:15

5aNS2. Limitations of predictions of noise-induced awakenings. Sanford Fidell (Fidell Assoc., Inc., 23139 Erwin St., Woodland Hills, CA 91367, sf@fidellassociates.com), Vincent Mestre (Landrum and Brown, Laguna Niguel, CA), Barbara Tabachnick (California State Univ., Emerita, Canoga Park, CA), and Linda Fidell (California State Univ., Emerita, Morro Bay, CA)

Awakenings attributable to transportation noise intrusions into residential sleeping quarters are surprisingly rare events. Current methods for estimating such awakenings from indoor sound exposure levels are problematic

for several reasons. They are based on sparse evidence and limited understandings; they fail to account for appreciable amounts of variance in dosage-response relationships; they are not freely generalizable from airport to airport; and predicted awakening rates do not differ significantly from zero over a wide range of sound exposure levels. Even in conjunction with additional predictors, such as time of night and assumed individual differences in "sensitivity to awakening," nominally SEL-based predictions of awakening rates remain of limited utility, and are easily mis-applied and mis-interpreted. Probabilities of awakening are more closely related to SELs scaled in units of standard deviates of local distributions of aircraft SELs, than to absolute sound levels. Self-selection of residential populations for tolerance of nighttime noise and habituation to airport noise environments offer more parsimonious and useful explanations for differences in awakening rates at disparate airports than assumed individual differences in sensitivity to awakening.

8:30

5aNS3. Determining annoyance thresholds of tones in noise. Jennifer Francis, Joonhee Lee, Adam Steinbach, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 5017 Underwood Ave #12, Omaha, NE 68132, jfrancis@unomaha.edu)

Mechanical equipment in buildings often produces noise with significant tones that can lead to complaints. Previous research has identified prominence levels of assorted tonal frequencies, but more work is needed to define at what level the tones at various frequencies lead to human annoyance. This project applies two different methods toward defining annoyance thresholds of tones in noise for two tonal frequencies: 125 Hz and 500 Hz. In the first, subjects are asked to perform a task, while exposed to ten minutes of a broadband noise spectrum with a tonal component set a certain level above the noise. They are subsequently asked to rate their annoyance to that noise condition. Five levels of each of the two tones are tested above two different background noise levels, 40 dBA and 55 dBA. In the second methodology, subjects adjust the level of the tone only above each of the two background noise levels until it is considered to be just annoying. Results obtained for the annoyance thresholds of tones in noise from each of these methods are compared.

8:45

5aNS4. Application of assorted tonality metrics to human annoyance thresholds of tones in noise. Joonhee Lee, Jennifer M. Francis, Adam Steinbach, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, jlee01@unomaha.edu)

Audible tones in background noise as produced by mechanical equipment in buildings can lead to complaints from occupants. A number of metrics including prominence ratio, tone to noise ratio, tonal audibility, and Aures' tonality have been developed to quantify the perception of tonalness, but it is not clear how these measures correlate with subjective annoyance response from listeners. This research investigates the relationship between tonality metrics and subjects' annoyance. As reported in another paper, two tests were conducted to determine annoyance thresholds for tones in noise. One involved having subjects rate their annoyance after being exposed to background noises with differing levels of tones while working on a given task. In the second test, subjects were asked to select the minimum tone level above a set background noise condition at which they began to feel annoyed. The thresholds according to assorted tonality metrics are calculated based on the subjective results and compared to previous research.

9:00

5aNS5. Listening as your neighbors: Simulation and evaluation of building structural transmission. Fangyu Ke, Cheng Shu (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY), Xiang Zhou (Home Entertainment Div., Bose Corp., Framingham, MA), Xuchen Yang, Gang Ren, and Mark Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu)

The enjoyment of music can be jeopardized by the knockings on the door and your furious neighbor shows up. Building structural transmission passes annoying sound to neighbors even when the audio is at low volume. In this paper, we implement a computational model for the simulation and evaluation of building structural transmission based on field measurements of structural transmission profiles and subjective listening tests. This model is intended to serve as a signal analysis tool that allows the audio engineers to evaluate the interference level using computer software and monitor system, so they can assess and mitigate these interferences without the need of conducting tedious subjective evaluation experiments in real acoustical environments. Our proposed transmission models are summarized from multiple acoustical measurements conducted in real room pairs to ensure its authenticity. Essentially we build "virtual" room pairs so we can simulate the interferences reaching the neighbors. We also implement a computational auditory model based on audio content analysis and subjective evaluation experiments to automatically evaluate the interference level caused by the transmitted sound. The proposed simulation and evaluation tool finds extensive applications in various areas of audio engineering such as audio content production, noise control, and audio system design.

9:15

5aNS6. Background noise levels on the fireground. Joelle I. Suits (Mech. Eng., Univ. of Texas, 204 E Dean Keaton St., Austin, TX 78712, jsuits@utexas.edu), Preston S. Wilson, and Ofodike A. Ezekoye (Mech. Eng. and Appl. Res. Labs., Univ. of Texas, Austin, TX)

An important part of signal detection and distinction is the strength of the signal compared to the levels of the background noise. On a fire scene, firefighters use a Personal Alert Safety System (PASS) device to locate trapped or injured personnel. When a firefighter becomes incapacitated, the device emits an audible alarm to help rescue teams locate the downed firefighter. This device has proven to be an invaluable part of a firefighter's equipment. However, there are cases in which the signal was not heard or localizable. It has become apparent that scientific research is necessary to improve this signal. The approach taken to investigating this environment is to use the passive sonar equation as a template. An important piece of this research is the background noise levels that are routinely found on an active fire scene. Much of the equipment used by firefighters acts as high intensity broadband noise sources. Measurements were taken to investigate the frequency, level content, and directionality of the equipment used on the fireground. The A-weighted spectrum of the equipment has then been compared to the signal emitted by a PASS device. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

9:30

5aNS7. Noise exposure in the general audience of a Formula 1 race. Craig N. Dolder, Joelle I. Suits, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu)

Formula 1 cars have a reputation for being among the loudest race cars. As such, noise exposure should be a major concern for not only drivers, crew and staff, but also audience members. Calibrated acoustic measurements were taken at multiple general admission areas at a Formula 1 race. Analysis of these measurements show that the exposure exceeded multiple standards for daily noise exposure limits within a fraction of the race. This presentation predicts the total noise exposure experienced at different track positions and what noise reduction rating would be required in order to reduce the exposure to safe levels. [This work was supported by Brüel & Kjær.]

9:45

5aNS8. A new class of personal noise monitoring devices in hearing conservation. David Yonovitz (Complex Data Systems, 2560 Via Pisa, Del Mar, CA 92014, jdata@sans.rr.com), Al Yonovitz (The Univ. of Montana, Missoula, MT), and Joshua Yonovitz (Yonovitz and Joe, LLP, Missoula, Montana)

In the United States, hearing loss affects approximately 35 million people. Occupational noise exposure, the most common cause of noise-induced hearing loss (NIHL) is nearly 100% preventable. Estimates of compensation for NIHL are estimated at 20 billion dollars, making NIHL the costliest environmental and medical-legal problem in the United States. Technology

may now be utilized to extend federal and state requirements for the preservation of hearing for those who are exposed to levels that may be injurious to hearing. While noise dosimetry has been routinely utilized to measure workplace exposure, the levels of noise exposure no longer should be predicted but actually measured from personal noise monitoring. Very compact devices with long battery life, high memory capacity to store over 3 months of daily exposure data, USB and automatic RF download capability are now available. The personal noise monitor records percentage of noise exposure integrated over a work day, decibel A-weighted levels, peak levels and the frequency spectrum of noise exposure. Signaling indicators to the wearer as well as acknowledgment of high exposure are part of the personal noise monitor. Trials in various work sectors and will be reported.

10:00–10:15 Break

10:15

5aNS9. Prediction of wind induced noise over bodies and small cavity. Huoy Thyng Yow, Dominic Hii, Anderson Saw, Cheah Heng Tan (Acoust. and Audio Signal Processing Group, Motorola Solutions Malaysia, Plot 2,Bayan Lepas Technoplex, Industrial Park, Mukim 12 SWD, Bayan Lepas, Pulau Pinang 11900, Malaysia, chy026@motorolasolutions.com), Ummi Masyitah Mohd Fisol, Zaidi Mohd Ripin, Norilmi Amilia Ismail, Abdullah Aziz Saad, Mohd Khairul Rabani Hashim, and Chan Ping Yi (School of Mech. Eng., Univ. Sains Malaysia Eng. Campus, Penang, Malaysia)

The problem with wind induced noise exists in any electronic device used in the outdoor that incorporates microphone components mounted underneath small opened cavities within its body. Due to the complexity nature of the problem, efforts in trying to experimentally understand the flow induced noise problem has lengthened product development cycle time. In this paper, a CFD approach using the SST k-omega turbulence model were utilized as an initial model to assist in understanding the phenomenon. Several instances of experimentation using actual prototype models were conducted for the correlation study. The experimentally perceived noise from the output of the microphone was found to be fundamentally correlated to the numerical analysis as evidenced from the flow velocity vectors, profiles, and vortex core inside the cavity region. It is also shown that by following the principle of nonlinear theory of cavity resonances and Rossiter's Theory, the understanding of the phenomenon can be further strengthened. Some inaccuracies in the numerical results stems largely from the choice of turbulence model and meshing configurations. Future work and activities required to extend and improve the current work were suggested at the end of this document.

10:30

5aNS10. Tonal noise sensitivity in hard drives. Matthew L. Nickerson and Kent Green (IBM, 3039 E. Cornwallis Rd., Durham, NC 27709, mlnicker@us.ibm.com)

High levels of noise have been found to cause offline events and data loss in information technology equipment in multiple data centers. Studies by this author and others showed these events were a result interaction between rotational storage media or hard drives (HDD) and air-borne noise. In particular, noise with a great degree of tonal content posed the greatest threat to HDD performance. A series of tests has been performed to understand this noise-HDD interaction. A test fixture was built and HDDs were exposed to sinusoidal signatures from 1000 Hz to 16 kHz. The incident sound pressure was monitored to capture the sensitivity of the HDDs at multiple sound pressure levels. Sound pressure, frequency and HDD performance were all correlated along with known HDD manufacturer provided vibratory sensitivities. Each subsequent generation of HDD showed greater incidence of tonal sensitivities and lower sound pressure level needed to achieve such high levels of performance degradation. Given the correlation with the vibratory sensitivities, the tonal response seems to be unique to the design and manufacture of each HDD individually.

10:45

5aNS11. Impedance assessment of an acoustic metamaterial-inspired acoustic liner. Benjamin Beck (NIA/NASA Langley, 500 Hosier St., Newport News, VA 23601, ben.beck@nasa.gov), Noah Schiller, and Michael Jones (Structural Acoust., NASA Langley Res. Ctr., Hampton, VA)

Acoustic liners are commonly used to reduce noise from commercial aircraft engines. Engine liners are placed in the nacelle inlet and aft bypass duct to attenuate the radiated noise from the fan, turbine, combustor, and other sources within the engine. Current engine liners are constructed of a metal perforate facesheet over a honeycomb structure. With this design, the low frequency performance of the liner is limited by the depth of the honeycomb. However, with advances in engine design, lower frequency performance is becoming more critical. Acoustic metamaterials can exhibit unique acoustic behavior using periodically arranged sub-wavelength resonators. Researchers have shown that metamaterials can effectively block the propagation of low-frequency acoustic waves. Therefore, acoustic metamaterial-inspired concepts are being investigated to improve the low frequency performance of engine liners. By combining the idea of a split-ring resonator metamaterial with a traditional quarter-wave acoustic liner, the low frequency acoustic absorption of the device can be significantly increased. The normal incident absorption coefficient of a metamaterial-inspired liner shows an increase from 0.05 to 0.8 at 600 Hz relative to a conventional honeycomb liner with the same depth while retaining the same 0.5 absorption coefficient at 1550 Hz.

11:00

5aNS12. Optimizing the acoustic performance of turbochargers. David Ledger, Dan Pruitt, and Paul Diemer (Preventative Acoust., BorgWarner Turbo Syst., 1849 Brevard Rd., Arden, NC 28704, dledger@borgwarner.com)

Turbocharger usage is predicted to increase in passenger vehicles for reasons such as enabling engine downsizing and improving thermodynamic efficiency. In order to achieve customer expectations, the acoustic performance of the turbo needs to be engineered for a wide range of operating load conditions. This requires the turbo supplier to set component targets in CAE and identify undesirable noises early in the development process using test cell acoustic measurements. Noise from a turbocharger can be separated into two main categories: (1) aeroacoustic sources including blade pass, pulsation, surge and broadband flow noise (2) structureborne from vibration sources such as subsynchronous, first order imbalance, and component resonances. One technique for comparing compressor acoustic performance is noise mapping, where the in duct noise data are collected across the full turbo operating range and post processed into 3D plots. Audible noises that can be generated in a centrifugal turbocharger and methods of identifying sources during development are presented in detail.

Session 5aPAa

Physical Acoustics: Advances in Infrasound Research III

Roger M. Waxler, Chair
NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677

Invited Papers

8:00

5aPAa1. Eigenray identification for non-planar propagation. Philp Blom and Roger Waxler (National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, psblom@olemiss.edu)

Identifying ray paths connecting a source and receiver at specific locations, or eigenrays, is a trivial exercise in the case of planar propagation since only the initial inclination angle and arrival range of the ray must be considered. However, in the case of propagation in three dimensions, the eigenray search becomes a non-trivial two parameter search. An iterative eigenray search method has been developed employing the auxiliary parameters used to solve the transport equation to efficiently obtain eigenrays connecting a source and receiver in three dimensions. An overview of the method will be presented along with applications to propagation in a simple model atmosphere and analysis of data obtained during the Humming Roadrunner experiment in the fall of 2012.

8:20

5aPAa2. Application of generalized autoregressive conditionally heteroskedastic modeling to wind noise for enhancing detection of transient infrasound. William G. Frazier and Greg Lyons (Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, frazier@olemiss.edu)

Autoregressive moving average (ARMA) process models are often used to represent stationary random processes. In applications where the process of interest is not stationary or has samples from a heavy-tailed distribution, extension to a generalized autoregressive conditionally heteroskedastic (GARCH) model can be used to improve the process representation. This modeling approach was developed by the financial industry and is frequently used in econometrics to represent financial time series possessing volatility clustering and to improve return forecasts. Recently, ARMA/GARCH models have been used to improve surface-to-surface radar detector performance where the clutter due to sea state possesses a heavy-tailed distribution. In this presentation, we will demonstrate the application of vector ARMA/GARCH models to the modeling of wind noise measured on infrasound arrays and to detection of transient infrasound in the presence of this noise.

8:40

5aPAa3. Measuring the coherency of microbaroms at long distances for the inversion of stratospheric winds. Omar Marcillo and Stephen Arrowsmith (LANL, P.O. Box 1663, Los Alamos, NM 87545, omarcillo@lanl.gov)

We demonstrate the design of an infrasound network and the associated analysis for measuring the coherency of microbaroms at large distances (10s of km) and inverting for stratospheric winds. We have developed a mathematical framework for the inversion of local stratospheric winds using microbaroms, and found constraints on the optimum sensor network topology. Based on these results, we deployed a prototype sensor network over the winter months (January to March, 2013) that comprised three single-sensor stations, one 30-m and two 1-km arrays with separations between 5 and 70 km. The initial analysis shows periods of very high coherency lasting several hours with tropospheric and low stratospheric celerities. Coherency decreases rapidly with distance and azimuth compared to the direction of propagation of microbaroms. We are exploring topography as the cause of low signal coherency at long distances. Following this pilot study, we are designing a denser sensor network further optimized to capture microbaroms and planning for its deployment and the validation of the inversion scheme using a Doppler Rayleigh Lidar system.

9:00

5aPAa4. Determining infrasound array performance from frequency response estimation. David Fee, Kit Dawson (Geophysical Inst., Univ. of Alaska Fairbanks, 903 Koyukuk Dr., Fairbanks, AK 99775, dfee@gi.alaska.edu), Roger Waxler (NCPA, Univ. of MS, University, MS), Curt Szuberla (Geophysical Inst., Univ. of Alaska Fairbanks, Fairbanks, AK), and Thomas Gabrielson (Appl. Res. Lab., Penn State Univ., State College, PA)

The performance of an infrasound array is typically described qualitatively. In this project, we begin determining quantitative metrics that define the performance of an infrasound array based upon changes in array parameter estimation, particularly direction-of-arrival and trace velocity. First, we determine the magnitude and frequency response of an International Monitoring System (IMS) infrasound array using the *in-situ* calibration technique. Phase differences in the frequency response between the reference and actual array elements are then represented as a shift in time. These times shifts are used to calculate time differences between non-redundant sensor pairs of the array and are then compared to theoretical time differences for incident plane waves. This allows us to estimate the

change in direction-of-arrival and trace velocity due to the unique frequency response of an array using a least-squares technique. We present detailed results for IMS arrays IS26, IS53, IS56, and IS57, which provides a good sampling of different sensors and wind-noise reduction systems in the IMS. For IS53 (Fairbanks, AK), the deviation in the azimuth estimation is bounded below 2 degrees at all frequencies of interest (>0.02 Hz). Preliminary results suggest array parameter estimation at other arrays can be significantly affected by changes in frequency response.

9:20

5aPAa5. Infrasound signal coherence across arrays: Observations from the international monitoring system. David Green (AWE Blacknest, Brimpton Common, Reading RG7 4RS, United Kingdom, dgreen@blacknest.gov.uk)

Arrays of microbarometers are designed to exploit the coherence of infrasound signals between sensors in order to detect and characterize the impinging wavefield. However, signal coherence decreases with increasing distance between measurement locations due to effects that include, but are not limited to, signal multi-pathing, dispersion, and wavefront distortion. Therefore, the design of an array is a balance between ensuring the sensor separations are small enough to guarantee acceptable signal coherence, yet large enough to provide the required resolution when estimating signal azimuth and velocity. Here, we report coherence measurements from more than 30 events that have been recorded on the International Monitoring System microbarometer arrays that are being constructed as one of the verification measures for the Comprehensive Nuclear-Test-Ban Treaty. The results confirm those of Mack and Flinn (1971) that signal coherence is greater in directions parallel, and is less in directions perpendicular, to the direction of propagation. In contrast to earlier studies, we report larger inter-event variation in coherence structure and provide an assessment of how these findings may influence future microbarometer array design.

9:40

5aPAa6. Interferometry applied to the large aperture infrasound array in the Netherlands. Julius Fricke, Läslo Evers (KNMI, PO Box 201, De Bilt 3730 AE, Netherlands, evers@knmi.nl), Kees Wapenaar (Dept. of GeoSci. and Eng., TU Delft, Delft, Netherlands), and Dick Simons (Acoust. Remote Sensing Group, TU Delft, Delft, Netherlands)

It has been theoretically shown by Wapenaar (2006) that the non-reciprocal Green's function can be retrieved with cross-correlation. Thus, interferometric techniques can be applied to a moving medium such as the atmosphere. Numerical experiments have shown that the delay times of stratospherically refracted infrasound can be obtained from cross-correlation between pairs of microbarometers. Doing so, information about the wind and temperature around the stratopause can be passively gathered from the stationary phase with, for example, the continuous noise of microbaroms. Actual measurements have been used from the Large Aperture Infrasound Array (LAIA) in the Netherlands. LAIA consists of several microbarometers with inter-station distances ranging from a few kilometers to tens of kilometers and is ideally suited to assess the results from theoretical and numerical experiments in practice. Results will be shown on the correlation length of infrasound from microbaroms and the effect of wind and temperature on the delay times retrieved from cross-correlations.

10:00–10:15 Break

Contributed Papers

10:15

5aPAa7. Beamforming methods in infrasonic array processing—Continuous signals. Philp Blom, Roger Waxler, and William Garth Frazier (National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, psblom@olemiss.edu)

It is often the case in analyzing infrasonic data that coherent "noise" due to natural and anthropomorphic sources obscures the phenomena of interest. Beamforming methods have been studied as a means of identifying continuous infrasonic signals and spatially removing them from a data record. An historical overview of frequency based array processing methods with be presented along with discussion of eigen-decomposition representations of various spatial spectral formulae. An application using spatial filtering to identify and extract microbaroms from an infrasonic data record will be demonstrated and discussed. Additionally, the framework for a robust, data-adaptive array processing method and the challenges associated with its implementation will be presented.

10:30

5aPAa8. Processing international monitoring system infrasound data to detect and locate global events using probabilistic algorithms. Stephen Arrowsmith, Omar Marcillo, George Randall (Los Alamos National Lab., 1711 Second St., Santa Fe, NM 87505, sarrowsmith@gmail.com)

Automating the detection and location of events using the International Monitoring (IMS) System infrasound network is a significant challenge. Any algorithm must reliably detect nuclear tests in the atmosphere with a minimum number of false alarms. Here, we report on the application of probabilistic techniques for detection, association, and location of infrasound events to data from the IMS network. We compare our results with the SEL3 automatic event detections obtained by the CTBTO.

10:45

5aPAa9. Consistent Wentzel-Kramers-Brillouin (WKB) approximation for acoustic-gravity waves in the atmosphere. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

The ray and WKB approximations have long been important tools of understanding and modeling propagation of atmospheric waves. However, contradictory claims regarding applicability and uniqueness of the WKB approximation persist in the literature. Here, linear acoustic-gravity waves (AGWs) in a layered atmosphere with horizontal winds are considered, and a self-consistent version of the WKB approximation is systematically derived from first principles and compared to ad hoc approximations proposed earlier. Parameters of the problem that need to be small to ensure validity of the WKB approximation are identified. Properties of low-order WKB approximations are discussed in some detail. Contrary to familiar cases of acoustic waves and internal gravity waves in the Boussinesq approximation, the WKB solution contains the geometric, or Berry, phase. Put differently, knowledge of the AGW dispersion relation is not sufficient for calculation of the wave phase. Significance of the Berry phase is illustrated by its effect on resonant frequencies of the atmosphere for vertically propagating waves.

11:00

5aPAa10. Reflection and transmission of low-frequency spherical waves at gas-liquid interfaces. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

High-frequency asymptotic solutions of the canonical problem of a spherical wave reflection at and transmission through a plane interface are well known. These asymptotics are usually derived using the stationary phase method and its extensions assuming that the distance between the point sound source and a receiver is large compared to the wavelength. The opposite case, where the distance is either of the order of or smaller than the wavelength, is of interest in a number of problems such as radiation of infrasound into the atmosphere by underwater explosions and earthquakes. Available quasi-stationary approximations fail to describe correctly acoustic power flux through the interface. Here, a low-frequency asymptotics of the acoustic field is obtained for gas-liquid interfaces. It is found that the acoustic field transmitted into gas can be approximately represented as a field due to a virtual point source in an unbounded medium. Positions of the virtual sources of the low- and high-frequency transmitted waves are compared. It is found that the bulk of acoustic energy transmission through the interface occurs at epicentral distances of the order of the source depth, regardless of the wavelength.

11:15

5aPAa11. Analysis of air-ground coupling using seismo-acoustic cross-spectral analysis. Robin S. Matoza (Scripps Inst. of Oceanogr., Univ. of California San Diego, IGPP 0225, La Jolla, CA 92093-0225, rmatoza@ucsd.edu) and David Fee (Geophysical Inst., Univ. of Alaska Fairbanks, Fairbanks, AK)

Air-ground and ground-air coupling are key processes in the rapidly developing field of seismo-acoustics and are particularly relevant in volcano geophysics. Volcanic eruptions often simultaneously generate sustained seismic and infrasonic signals, e.g., by fluid flow simultaneously occurring in the subsurface and atmosphere. Building upon recent work by Ichihara *et al*. (2012), we show that cross correlation, coherence, and cross-phase spectra between waveforms from nearly collocated seismic and infrasonic sensors provide new information about air-ground and ground-air coupling. Combining this method with infrasound array processing can provide insight into the geophysical processes generating a range of seismo-acoustic signals. For example, we show that seismic tremor recorded during an eruption at Mount St. Helens is dominated by air-ground coupled infrasound between 5 and 15 Hz. We anticipate that cross-spectral analysis and similar techniques will have wide applicability to arbitrary seismo-acoustic sources and in exploiting the growing volume of seismo-acoustic data. Ichihara *et al*. (2012), “Monitoring volcanic activity using correlation patterns between infrasound and ground motion,” Geophys. Res. Lett. **39**, L04304, doi:10.1029/2011GL050542.

Session 5aPAb

Physical Acoustics: Topics in Physical Acoustics

Bradley Goodwiller, Cochair
Mech. Eng., The Univ. of Mississippi, 1 Coliseum Dr., Rm. 1079, University, MS 38677

Tiffany Grey, Cochair
National Ctr. for Physical Acoust., 1 Coliseum Dr., University, MS 38677

Contributed Papers

8:00

5aPAb1. Multiple methods for calculating ultrasonic attenuation in liquids from non-invasive swept frequency experiments. Anirban Chaudhuri and Dipen N. Sinha (Sensors and Electrochemical Devices (MPA-11), Los Alamos National Lab., P.O. Box 1663, M.S. D429, Los Alamos, NM 87545, anirban@lanl.gov)

The swept frequency acoustic interferometry (SFAI) technique determines multiple acoustic properties of fluids in a single broadband frequency sweep measurement in a noninvasive manner. The technique employs two ultrasonic transducers on the outside of a test cell containing the liquid, one to transmit a swept-frequency signal and the other to receive the transmitted signal. Several different methods to extract frequency-dependent ultrasonic attenuation in liquids from the broadband frequency response (resonance spectrum) of the composite test cell will be described in this paper. These approaches take advantage of the characteristics of both the frequency response and the time response (derived from the Fourier transform) of the experimental data. These methods were first verified with simulated SFAI data generated using a 1-D multi-layer transmission model; a brief description of the model is presented. Attenuation values were then calculated for different fluids (deionized water, acetone, Fluorinert, 10W-30 motor oil, and crude oil) from experimental swept frequency data obtained using a specially designed laboratory test cell. The attenuation values computed using the different methods are consistent with each other, as expected. However, based on the quality of the experimental data, certain methods may be more appropriate in a given measurement condition.

8:15

5aPAb2. Numerical computation of acoustic radiation force in two and three dimensional resonators. Ari Mercado (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Box S-5024, Springfield, MA 01119, blipkens@wne.edu), Brian McCarthy, Ben Ross-Johnsrud, Jason Dionne (FloDesign Sonics, Wilbraham, MA), and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

Large scale acoustophoretic separation of particles in a flow field can be accomplished by trapping the particles, i.e., remain in a stationary position, in ultrasonic standing waves. The particles are collected in a column pattern, separated by half a wavelength. Within each nodal plane, the particles are trapped in the minima of the acoustic radiation potential. The axial component of the acoustic radiation force drives the particles to their stable axial position. The radial or lateral component of the acoustic radiation force is the force that traps the particle. It must be larger than the combined effect of fluid drag force, i.e., Stokes drag, and gravitational force. There is a need for a better understanding of the lateral acoustic radiation force in realistic acoustophoretic separation devices. COMSOL Multiphysics® software was used to predict the acoustic field in two and three dimensional models of acoustophoretic separation devices driven by piezoelectric transducers. The resulting acoustic field was then used to calculate the acoustic radiation force acting on a suspended particle in two and three dimensions by applying Gor'kov's equation. Measurements of trapped particles in standing waves indicate accurate calculations of acoustic field and radiation force. [Work supported by NSF PFI:BIC 1237723.]

8:30

5aPAb3. A kinematic shaker system for dynamic stabilization of Rayleigh-Bénard convection. Anand Swaminathan (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, azs5363@psu.edu), Matthew E. Poese, Robert W. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA), and Steven L. Garrett (Graduate Program in Acoust., The Penn State Univ., State College, PA)

The ability to dynamically inhibit Rayleigh-Bénard convection using acceleration modulation is of interest to groups who design and study thermoacoustic machines, as the introduction of unwanted convection can have deleterious effects on the desired operation and efficiency of the device. These performance issues, caused by suspected convective instability, have been seen both in traveling wave thermoacoustic refrigerators and in cryogenic pulse tube chillers [Swift and Backhaus, J. Acoust. Soc. Am. **126**, 2273 (2009)]. This presentation discusses the vibratory conditions under which a small, rectangular container of statically unstable fluid may be stabilized by vertical vibration, applying the computational methods of Carbo [Ph.D. Thesis, Penn State Univ. (2012)]. The required shaking velocities for stabilization are found to be too large for an electrodynamic shaker. As a solution, a shaker system employing a kinematic linkage to two counter-rotating flywheels, driven by a variable-speed electrical motor, produces peak-to-peak displacements of 15.24 cm to a platform mounted on two guide rails. Thus far, this shaker has produced peak speeds of up to 3.7 m/s. [Work supported by the Office of Naval Research, the ARL Walker Graduate Assistantship, and ARPA-E.]

8:45

5aPAb4. Investigation of sound diffraction in periodic nano-structures using acoustic microscopy. Anurupa Shaw (Georgia Tech Lorraine, 2 Rue Marconi, Metz- Technopole 57070, France, shaw.anurupa@gmail.com), Suk Wang Yoon (Georgia Tech-CNRS UMI2958, Georgia Tech Lorraine, Metz-Technopole, France), and Nico F. Declercq (Georgia Inst. of Technol., George W. Woodruff School of Mech. Eng.,Lab. for Ultrasonic Destructive Evaluation, Georgia Tech-CNRS UMI2958,Georgia Tech Lorraine, Metz-Technopole, France)

Acoustic microscopy is a well established technique as far as smooth surfaces are concerned. Traditionally V(z) curves are obtained from which, through surface wave generation, important features concerning elasticity and related properties can be extracted. More recently, high resolution imaging based on acoustic microscopy has appeared. In this study, we investigate the possibility to investigate samples with nano-structures.The surface profile of the samples have periodic structures but lack smoothness. The periodicity causes sound diffraction and the roughness influences the acoustic microscopic investigation. Small acoustic contrast between the substrate

and the periodic corrugation materials give us information about the additional stresses which develop and affect the bonding between the two materials. In this presentation, a description is presented of the experiments and a comparison is made between results on smooth surfaces and results on periodic structures of the same material. An attempt is made to analyze the effects described above.

9:00

5aPAb5. Acoustic wood anomaly phenomenon in transmission and diffraction fields. Jingfei Liu (Georgia Inst. of Technol., 2, rue Marconi, Metz 57070, France, benjamin.jf.liu@gatech.edu) and Nico F. Declercq (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Metz, France)

The notion of “Wood anomaly” in diffraction spectra in acoustics has been mostly investigated in normal reflection fields obtained from periodically corrugated interfaces between a solid and a liquid. In this work, the phenomenon is investigated both in transmission field as well as in reflection, both normal and onblique. Experiments are performed on a brass-water interface and also on a water-brass interface to obtain the reflection from these interfaces and the transmission through the interfaces. It is observed that the phenomenon appears not only in reflection but also in transmission, however it occurs at different frequencies. The study focuses on the physical origins of these phenomena and from the point of view of different possible explanations such as surface acoustic wave generation and local resonance of the corrugation.

9:15

5aPAb6. Nonlinear frequency mixing, bubbly liquids, and standing waves: A numerical study. Christian Vanhille (Universidad Rey Juan Carlos, Tulipán s/n, Móstoles 28933, Spain, christian.vanhille@urjc.es), Cleofé Campos-Pozuelo (Consejo Superior de Investigaciones Científicas, Madrid, Spain), and Dipen N. Sinha (Los Alamos National Lab., Los Alamos, NM)

We study the nonlinear frequency mixing in a resonator in which a bubbly liquid (nonlinear medium) is considered. The acoustic pressure source at one end of the cavity excites the bubbly liquid at two frequencies. The cavity is set to be resonant at the difference frequency in the bubbly liquid. Numerical simulations are carried out at high and small amplitudes. Results show the formation of a difference frequency and one higher harmonic at high amplitude. [This work is part of the research project DPI2012-34613 (Spain).]

9:30

5aPAb7. Influence of texture anisotropy on acoustoelastic birefringence in stressed wood. Jinxia Liu, Jianping Zhou, Songming Qi, Zhiwen Cui, and Kexie Wang (College of Phys., Jilin Univ., qianjin2699,Jilin Univ, Changchun 130012, China, jinxia@jlu.edu.cn)

The velocities of shear waves propagating transversely to the applied stress direction in specimen were investigated to study the effect of anisotropy on acoustoelastic birefringence for birch and elmwood specimens. The oscillation direction of the shear waves with respect to the wood plate axis was varied by rotating an ultrasonic sensor, and the relationship between the shear wave velocity and the oscillation direction was examined. The ultrasonic velocity of shear waves propagated through radial direction of wood plate specimen, transversely to the loading direction. The results indicated that when the oscillation direction of the shear wave corresponded to one of the anisotropic directions of the wood plate specimen, the shear wave velocity decreased sharply and the relationship between shear wave velocity and rotation angle tended to become discontinuous. The azimuth angles of extremum value mainly depended on the texture anisotropy and were almost not influenced by stress-induced anisotropy. Shear wave polarization preferentially followed the direction of texture anisotropy. The results of both birch and elmwood specimens showed that shear wave velocity slightly increased with compressive stress at almost all rotated angles and exhibited slight acoustoelastic birefringence effects.

FRIDAY MORNING, 6 DECEMBER 2013 PLAZA A, 8:00 A.M. TO 12:00 NOON

Session 5aPP

Psychological and Physiological Acoustics: Hello, Hello... is Anyone Listening? (Poster Session)

Suzanne Levy, Chair
EarLens Corp., 200 Chesapeake Dr., Redwood City, CA 94063-4745

Contributed Papers

5aPP1. System for automatic personalization of head-related transfer functions based on computer vision, photo-anthropometry, and inference from a database. Edgar A. Torres Gallegos, Felipe Orduña-Bustamante, and Fernando Arámbula-Cosío (Centro de Ciencias Aplicadas y Desarrollo Tecnológico, Universidad Nacional Autónoma de México, Circuito Exterior S/N, Ciudad Universitaria. A.P. 70-186, México, D.F., Distrito Federal 04510, Mexico, edgar.augusto.torres.gallegos@gmail.com)

A software system and method for automatic personalization of head related transfer functions (HRTF) is presented. The system pursues the objective of personalizing HRTF using the anthropometry of the subject, as proposed originally by [Zotkin *et al.* WASPAA 2003, 157–160], which in this work, is measured automatically from a set of photographs of the subject. The system operates in three stages. First, a computer vision algorithm, known as active shape models, is used over the portraits to recognize and adjust geometric form profiles of the ears, head, and torso of the subject. Then, anthropometry is performed by measurement of pixel distances between specific points in the model, and then converted into metric units. Finally, HRTF are estimated from the anthropometry using a choice of three methods, whose performance is compared: (1) HRTF selection of the best anthropometric match in the CIPIC database, (2) HRTF synthesis by multiple linear regression and principal component analysis, and (3) HRTF synthesis using an artificial neural network. Results are analyzed, concluding that automatic personalization of HRTF is attainable automatically from the subject portraits, using computer vision and inference from a database. Further analysis, however, reveals the need for more complete HRTF and anthropometric databases.

5aPP2. A phon loudness model quantifying middle ear and cochlear sound compression. Julius Goldstein (Hearing Emulations LLC, 882 Page Ave., St. Louis, MO 63114-6105, goldstein.jl@sbcglobal.net)

Phon loudness data by Lydolf and Møller [1997, see Suzuki and Takeshima, J. Acoust. Soc. Am. **116**, 918 (2004)] at third octave frequencies 20–1000 Hz suggest evidence for Middle-Ear (ME) Sound Compression (MESC) at loudness above 60 phons. Parameters of a phon model based on sound compression by an idealized cochlear amplifier (Goldstein, 2009, 2011) were fit to the phon data at low sound levels. MESC was estimated as the excess in sound pressure levels of phon data over model predictions for 80–100 phons. MESC was found when cochlear amplifier inputs exceeded the known Acoustic Reflex Threshold (ART) of ~83 dB SPL, causing ME Muscle (MEM) contraction. Physiologically consistent first-order highpass ME transfer functions reveal a mix of MEM attenuation and attenuation at 20–100 Hz attributed to ME static pressure from pressure-field stimuli >100 dB SPL. These attenuations are isolated with the ME modeled as the low side of an RLC analog bandpass. The original data above 60 phons are modeled accurately as the sum of the two attenuations, the cochlear amplifier input, and normal ME attenuation re 1 kHz. The considerable basic information estimated from the phon data can be compared with other sources to guide further integrative research. [NIH funded 1972-04.]

5aPP3. Acoustics-structure interactions in the human middle ear produce variety of motion modes at the malleus-incus complex. Hongxue Cai, Ryan P. Jackson, Charles R. Steele, and Sunil Puria (Mech. Eng., Stanford Univ., 496 Lomita Mall, Stanford, CA 94305, hongxuec@stanford.edu)

We developed a 3D finite-element model to simulate the dynamics of the human middle ear, using COMSOL Multiphysics software to solve the resulting acoustics-structure interaction problem. We validated the model by comparing numerical results with experimental data measured in the ear canal, on the tympanic membrane (TM), at the umbo, and at the stapes footplate. The results show that at low frequencies (up to 500 Hz), the conventionally accepted hinge-like motion of the malleus-incus complex dominates the response, with the angle between the rotational axes of the malleus and incus staying below about 5 degrees. However, above 5 kHz, this angle becomes significantly larger, indicating that the malleus and incus rotate about different axes. Near the upper frequency limit of 20 kHz, the angle between the rotational axes of the malleus and incus approaches 90 degrees as the malleus adopts a lower-inertia twisting-like rotation about its first principal axis. The model is also used to explore the effects, on ossicular motion and overall pressure transfer from the ear canal to the cochlea, of alterations to the mechanical properties of the TM, to the flexibility of the malleus-incus joint, and to the mass of the ossicles. [Funded by NIDCD/NIH.]

5aPP4. Spatial separation decreases psychoacoustic roughness of high-frequency tones. Julián Villegas (Comput. Arts Lab, Univ. of Aizu, The University of Aizu, Tsuruga, ikki-machi, Aizu Wakamtsu, Fukushima 965-8580, Japan, julian@u-aizu.ac.jp), William L. Martens (Faculty of Architecture, Design and Planning, Univ. of Sydney, Sydney, NSW, Australia), Michael Cohen (Comput. Arts Lab, Univ. of Aizu, Aizu Wakamatsu, Japan), and Ian Wilson (CLR Phonet. Lab, Univ. of Aizu, Aizu Wakamatsu, Japan)

Perceived roughness reports were collected for pairings of sinusoidal tones presented either over loudspeakers or headphones such that the sounds were collocated or spatially separated 90 degrees in front of the listener (+/− 45 degrees). In the loudspeaker experiment, pairs of sinusoids were centered at 0.3, 1.0, and 3.3 kHz, and separated by half a critical band. In the headphone experiment, the pairs of sinusoids were centered at 0.5, 1.0, and 2.0 kHz, and separated by a semitone. Although not all listeners' reports showed the influence of spatial separation as clearly as others, analysis indicates that listeners generally found spatially separated tone combinations less rough when the frequencies of those tones were centered at 2.0 kHz or higher. This trend was also observed in a follow-up study with 20-component complex tones at fundamental frequencies of C2, C3, A4, and C4 (131, 262, 440, and 523 Hz, respectively) presented via headphones. These results suggest that spatial separation decreases perceived roughness, especially for tones with frequencies higher than the threshold at which interaural time differences rival interaural level differences for sound localization (approximately 2.3 kHz) and that the current roughness models need to be reviewed to include binaural effects.

5aPP5. Microstructure of auditory sensitivity within audiometric frequencies. Rita Quigley and Al Yonovitz (The Univ. of Montana, 32 Campus Dr., Missoula, MT 59812, rita.quigley@mso.umt.edu)

Variations in audiometric thresholds between standard audiometric test frequencies may be related to speech discrimination and cochlear "dead regions." Twenty-four logarithmically spaced frequencies between octaves were examined. A computer controlled the presentation of signals and a subject responded using a unique threshold testing algorithm that was designed to minimize the test time. Patterns of hearing loss were compared to normal variations. The hearing loss patterns included a flat loss, a noise-induced loss and a high frequency loss. The pattern of this threshold audiogram and the comparison to a test of cochlear dead regions will be discussed.

5aPP6. The effect of firefighter personal protective equipment on auditory thresholds. Joelle I. Suits (Mech. Eng., Univ. of Texas, 204 E Dean Keaton St., Austin, TX 78712, jsuits@utexas.edu), Craig A. Champlin (Commun. Sci. and Disord., Univ. of Texas, Austin, TX), Preston S. Wilson, and Ofodike A. Ezekoye (Mech. Eng. and Appl. Res. Labs., Univ. of Texas, Austin, TX)

Communication on a fire scene is essential to the safety of firefighters. Not only to be able to hear and understand radio chatter, but also alarm signals used on the fireground. One such alarm is the Personal Alert Safety System (PASS) device. This device is used to help locate a downed firefighter. One part of this complex problem is the effect of the protective equipment (helmet, eye protection, hood, coat) on hearing. Previous findings have shown the effect of this protective equipment on head related transfer functions using a KEMAR. [Suits *et al*. (2013, June). Paper presented at the International Congress on Acoustics, Montreal, Canada] The physical acoustic measurements showed a change in the signal that would reach the tympanic membrane. To relate the findings of the physical measurements to human reactions, the change in auditory threshold caused by wearing the personal protective equipment was measured. The changes seen in the physical acoustics measurements caused the auditory threshold of the subjects to increase at higher frequencies. The measured increases at 3000 Hz, 4000 Hz, and with an example PASS signal were between 5 and 10 dB. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

5aPP7. Differences elicited by intensity changes on non-monotinicities observed in amplitude and quasi-frequency modulation discrimination. Ewa Borucki and Bruce G. Berg (Cognit. Sci., UC Irvine, 2201 Social & Behavioral Sci. Gateway Bldg., Irvine, CA 92697-5100, eborucki@uci.edu)

This study investigated the influence of intensity on the effects of cubic distortion tones (CDTs) in a task traditionally used to investigate the bandwidths of phase sensitivity. For a 2000 Hz carrier, estimates of modulation depth necessary to discriminate amplitude modulated (AM) tones and quasi-frequency modulated (QFM) were measured in a two interval forced choice task as a function modulation frequency. Threshold functions for the listeners were nonmonotonic, with sharp nonmonotonicities observed at modulation frequencies above 300 Hz. This is likely to be due to the generation of a CDT at the site of the lower sideband, creating a salient spectral cue. When stimulus intensity is decreased from 80 dB to 40 dB, a greater number of non-monotonicities are observed for high modulation frequency conditions. The decrease in intensity appears to create phasic differences in the CDT altering threshold functions.

5aPP8. Statistical structure of speech sound classes is congruent with cochlear nucleus response properties. Christian Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., University of Louisville, Louisville, KY 40292, christian.stilp@gmail.com) and Michael Lewicki (Elec. Eng. and Comput. Sci., Case Western Reserve Univ., Cleveland, OH)

Natural sounds possess considerable statistical structure. Lewicki (2002, *Nature Neurosci*. ***5(4)***, 356–363) used independent components analysis (ICA) to reveal the statistical structure of environmental sounds, animal vocalizations, and human speech. Each sound class exhibited distinct statistical properties, but filters that optimally encoded speech closely resembled response properties in the mammalian auditory nerve. This and other analyses of statistical properties of speech examine only global structure without considering systematic variability in different speech sound classes, while acoustic/phonetic analyses of these classes are agnostic to their statistical structure. Here, statistical structure was investigated in principled subdivisions of speech: consonants organized by manner of articulation, and vowels organized by vocal tract configuration. Analyses reveal systematic differences for local statistical structure in speech: statistically optimal filters in ICA were highly diverse for different consonant classes but broadly similar for different vowel classes. While the global statistical structure of speech reflects auditory nerve response properties (Lewicki, 2002), local statistical structure of speech sound classes is well-aligned with cochlear nucleus response properties. Results support theories of efficient coding, in which sensory systems adapt and evolve in order to efficiently capture natural stimulus statistics.

5aPP9. Time-series evaluation of the sense of presence in audio content. Kenji Ozawa, Shota Tsukahara, Yuichiro Kinoshita, and Masanori Morise (Dept. of Comput. Sci. and Eng., Faculty of Eng., Univ. of Yamanashi, 4-3-11 Takeda, Kofu, Yamanashi 400-8511, Japan, ozawa@yamanashi.ac.jp)

The authors have been making an effort to clarify the property of the sense of presence because this sense can be crucial for evaluating audio equipments. In our previous studies, the overall evaluation of presence was conducted for a set of audio content items, i.e., the subjects were asked to evaluate the presence of a whole content item. In this study, time-series evaluation of the sense of presence was conducted using the method of continuous judgment by category. Stimuli were 40 binaural content items with duration of approximately 30 s. They were recorded with a dummy head and presented to subjects via headphones. Twenty subjects judged the sense of presence for every moment during the presentation of each item using seven categories by pressing one of seven keys on a keyboard. After the time-series evaluation, the subjects also evaluated the overall presence of the item by seven categories. The results showed that the overall presence was highly correlated with the five-percentile exceeded presence level, which is the level exceeded for the 5% of the time under consideration. Moreover, the latency of the evaluation tended to be longer when the presence of an item was low. [Work supported by NICT.]

5aPP10. Less than careful speech: Exploring the roles of target duration and time varying intensity in spoken language processing. Ryan G. Podlubny and Benjamin V. Tucker (Linguist., Univ. of Alberta, 11323 110th Ave., Edmonton, AB, Canada, podlubny@ualberta.ca)

Despite substantial phonetic variation in the production of natural speech, listeners are quite adept at processing varying speech signals. Further, the recognition of spontaneous speech is a multifaceted problem where listeners are drawing information from many sources in the recognition process. Previous research has begun to outline the influence of both syntactic and semantic contributions to spoken word recognition; however, it has been argued (Ernestus *et al*., 2002; van de Ven *et al*., 2012) that such cues reach a ceiling in their contributions, and that acoustic information also aids the listener when processing language. There has been limited discussion on what information encoded within a given speech signal lends itself to more effectively processing spontaneous speech. The present study contributes to such a discussion, using responses from 85 participants in modified Cloze tasks and speech in noise manipulations, where results serve as a starting place to describe the contributions of duration and intensity in natural speech (as well as their relative effects on spoken language processing). The introduction of particular acoustic information—over and above syntactic and semantic cues—has been observed within this study as a statistically significant contributor to more effective spoken language processing.

5aPP11. Effects of linguistic background and noise on perception of fricatives. Megan T. Stevens, Harisadhan Patra, and Petula C. Vaz (Audiol. & Speech Pathol., Bloomsburg Univ. of PA, 226 CEH, 400 E 2nd St., Bloomsburg, PA 17815, hpatra@bloomu.edu)

This study examined how native American English (AE) speakers perceived fricatives spoken by native AE and Bangladeshi Bengali (BB) speakers in quiet and noise. Participants included seven normal-hearing adults between 20 to 26 years of age. Participants listened to speech tokens of five fricatives, /s/, /z/, /ʃ/, /f/, and /v/ in the initial, medial, and final positions in the context of the point vowels /a/, /u/, /i/. Multitalker babble (MTB), speech noise, and three narrow bands of noise, 1000–2000 Hz, 2000–4000 Hz, and 500–5000 Hz at 45 dB SPL, 65 dB SPL and 85 dB SPL were used. The results suggested that listeners perceived fricatives significantly better when spoken by AE compared to BB speakers, and in quiet than in noise, especially in MTB. Listeners had the most difficulty with /z/, followed by /s/, /v/, /f, and /ʃ/ respectively, when tokens were spoken by BB speakers. This study may have implications for accent reduction therapy as well as for teaching English to English-language learners, especially when the phonology of the learners' native language differs greatly from that of AE. Further studies are warranted especially due to an increasingly growing non-AE speaking population in the United States.

5aPP12. Effects of noise and speaker's language background on plosive perception. Julie A. Brent, Harisadhan Patra, and Petula C. Vaz (Audiol. & Speech Pathol., Bloomsburg Univ. of PA, 226 CEH, 400 E 2nd St., Bloomsburg, PA 17815, hpatra@bloomu.edu)

This study examined the effect of speakers' language background and noise on the perception of American English (AE) plosives. Six normal-hearing, young-adults volunteered for the study. Participants listened to speech tokens of six plosives, /p/, /b/, /t/, /d/, /k/, and /g/ in the initial, medial, and final positions in the context of three vowels, /a/, /i/, and /u/, spoken by native Bangladeshi Bengali (BB) and AE speakers. Tokens were presented at 45, 65, and 65 dB SPL, either in quiet or noise. Multitalker babble, speech noise, and 1000–2000 Hz, 2000–4000 Hz, and 500–5000 Hz noise bands were used as noise. Significant differences were found with all noise types across both language backgrounds; the most difficult noise type condition being the multitalker babble. Listeners performed the best at 65 dB SPL. Listeners perceived plosives spoken by AE than BB speakers with significantly greater accuracy, more so in noise, except in the final positions. For BB-spoken tokens, listeners had the most difficulty with voicing and aspiration features. Voiceless sounds were easily confused with voiced, especially in the initial positions. This study has significant implications for accent reduction therapy as well as for teaching English to English Language Learner's. Further studies are warranted.

5aPP13. Predicting the speech reception thresholds with physical metrics. Fei Chen and Lena L. N. Wong (Div. of Speech and Hearing Sci., The Univ. of Hong Kong, Rm. 546, Prince Philip Dental Hospital, 34 Hospital Rd., Sai Ying Pun, Hong Kong, Hong Kong, feichen1@hku.hk)

Many measures [e.g., speech transmission index (STI) and speech intelligibility index (SII)] have been proposed to predict the speech intelligibility in noise. Nevertheless, most of these studies were performed under the conditions with a limited number of maskers. The present study further investigated how well the present speech intelligibility and quality metrics predicted the speech reception thresholds (SRTs) for sentences corrupted by stationary and fluctuating maskers. The SRT scores were collected from 30 normal-hearing (NH) and 15 hearing-impaired (HI) native-Cantonese listeners. Sentences were corrupted by nine types of maskers, including speech-shaped noise and eight real-life environmental noises (4- and 6-talker babbles, upper and lower deck in bus, cafe, Chinese restaurant, MTR carriage, and street). The resulting average SRT scores were subject to the correlation analysis with various metrics computed from the noise-masked sentences. Of all the objective metrics considered, the STI and CSII measures

performed the best, and their high correlations (i.e., r = 0.91 to 0.96) were maintained in both NH and HI conditions. This suggests that some of the physical metrics that have been found previously to correlate highly with the intelligibility of sentences in noise may also be used to predict the SRTs affected by different maskers.

5aPP14. Disyllabic Mandarin lexical tone perception by native Dutch speakers: A case of adult perceptual asymmetry. Christian Hoffmann, Makiko Sadakata (Ctr. for Cognition, Radboud Universiteit Nijmegen, Donders Inst. for Brain, Cognition & Behavior, Montessorilaan 3, Nijmegen 6525HR, Netherlands, c.hoffmann@donders.ru.nl), Ao Chen (Utrecht Inst. for Linguist, Universiteit Utrecht, Utrecht, Netherlands), and James M. McQueen (Donders Inst. for Brain, Cognition & Behavior, Ctr. for Cognition, Radboud Universiteit Nijmegen, Behavioral Sci. Inst., Nijmegen, Netherlands)

Asymmetries in the perception of lexical tones have previously been reported in infant studies: the order in which certain tones are presented influences discrimination accuracy. Using disyllabic materials, the current study provides evidence that such asymmetries can be found for L2 learners of Mandarin Chinese as well, making their responses in identification and discrimination tasks qualitatively different from adult native speakers. Using an active oddball paradigm, we found that Tone 4 deviant within a Tone 1 standard environment was consistently easier to discriminate than the reverse condition. Furthermore, this difference is also reflected in amplitude differences of the auditory N2/P3 complex. In two subsequent EEG experiments, we systematically varied (a) the relative acoustic difference between standards and deviants and (b) stimulus variance within standards and deviants. Results indicated that both decreased acoustic difference as well as increased stimulus variance positively increase relative perceptual asymmetry as well as the relative difference between ERP responses. Finally, we compare multiple mechanisms by which the native/non-native pitch system might cause these perceptual asymmetries.

5aPP15. Perceptual contribution of vowel sub-segments to Mandarin tone identification. Fei Chen, Lena L. N. Wong, and Eva Y. W. Wong (Div. of Speech and Hearing Sci., The Univ. of Hong Kong, Rm 546, Prince Philip Dental Hospital, 34 Hospital Rd., Sai Ying Pun, Hong Kong, Hong Kong, feichen1@hku.hk)

Recent noise-replacement studies showed that (1) vowels carried more intelligibility information in Mandarin sentence recognition, and (2) a little vowel onset portion could significantly increase the intelligibility when it was added to the consonant-only Mandarin sentences. This study further evaluated the perceptual contribution of vowel sub-segments to Mandarin tone identification. The original duration-normalized vowels (FULL) were modified to produce two types of stimulus, i.e., (1) Left-only [LO (p)], which preserved p = 10% to 50% of the initial vowel portion, and replaced the rest vowel portion with speech-shaped noise (SSN), and (2) center-only [CO (p)], which preserved p = 15% to 60% of the center vowel portion, and replaced the rest initial and final vowel portions with SSN. Tone identification scores were collected from 20 normal-hearing native-Mandarin listeners. Results in the present study showed that (1) Mandarin tone perception at the LO (10%) condition was slightly higher than the chance level (i.e., 25%); (2) tone identification at the CO (60%) condition was not significantly different with that at the FULL condition. These findings suggest that vowel onset portion provides information redundant to vowel centers for Mandarin tone identification, and vowel centers contain sufficient information for reliable Mandarin tone identification.

5aPP16. Phonemic word memorization strategy between second language learners and their native speakers analyzed by Rey's Auditory Verbal Learning Test. Keiko Asano (School of Medicine, Juntendo Univ., 1-11-2905, Kinko-cho, Kanagawa-ku, Yokohama-City 221-0056, Japan, kiasano@uu.em-net.ne.jp)

In order to increase the numbers of vocabulary, it is effective way for L2 learners to acquire the memorization strategies. This study investigated what kinds of strategies L2 learners of English and their Native speakers of Japanese use in order to memorize and retrieve words aspects of Rey's Auditory Verbal Learning Test (AVLT). This Auditory Verbal Leaning Test is widely spread to examine word learning and memory in the field of clinical and neuropsychological assessments. This test has two parts of stages: A list of 15 unrelated, concrete nouns is presented over three learning trials with immediate recall tested following each presentation. Next, after a delay interval of some 10 minutes with no further presentations of the list, delayed recall is assessed. The number of words correctly recalled is commonly adopted as quantitative information in clinical assessment. In addition to these procedures, after the tests participants are asked what kinds of strategies they used to memorize the words in terms of examining the process of decoding, recall and retrieval of words. Self-monitoring of memorization strategies by L2 learners of English are tended to use the phonemic or phonetic-oriented strategies whereas their Native speakers of Japanese use rather visual-oriented and episode-making. The implication on the function of different brain area activated between L2 learners and Native speakers will also be discussed.

5aPP17. Selective and divided attention: Spatial and pitch "spotlights" in a non-semantic task. Lindsey Kishline, Eric Larson, Ross K. Maddox (Inst. of Learning and Brain Sci., Univ. of Washington, 1715 Columbia Rd. NE, Portage Bay Bldg., Rm. 206, Seattle, WA 98195, l.kishline@gmail.com), and Adrian KC Lee (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Listeners can reliably attend to one auditory object in the presence of many, but how good are they at dividing their attention among multiple auditory objects in a crowded auditory environment? Previously, divided attention has been looked at under the "spotlight" model of attention, borrowed from vision research, which predicts enhancement of specific spatial regions. However, the details of how these spotlights are deployed remain unknown. Here we used six competing auditory objects (distributed in azimuth in one experiment and in pitch in the other) and asked listeners to attend some and ignore others. Whether or not the target objects were contiguous tested whether they employed a single spotlight of varying size, or multiple spotlights with varying degrees of separation. Results suggest that listeners can reliably attend to multiple auditory objects. However, the level of accuracy was dependent upon the number of attended targets and their configuration.

5aPP18. Dynamic component analysis for multi-input/multi-output problems, with application to speech and neurophysiology. Erik Edwards and Edward F. Chang (Dept. of Neurological Surgery, UC San Francisco, Sandler Neurosci. Bldg., 675 Nelson Rising Ln., 535, San Francisco, CA 94143, erik.edwards4@gmail.com)

We explore a set of methods referred to collectively as dynamic component analysis (DCA) to derive dictionaries of dynamical patterns for speech (TIMIT database). The methods use spatio-temporal singular value decomposition (ST-SVD) and common spatio-temporal pattern (CSTP) matrix computations. When used on the speech spectrogram, these yield a transformation to a new set of time series representing extracted features at reduced bandwidth. The method is computationally efficient (closed-form solutions suitable for real-time) and robust to additive noise, with diverse applications in speech processing and general multi-input/multi-output (MI/MO) modeling. When used to predict a single neural output (MI/SO), this gives an efficient new method for deriving the spectro-temporal receptive field (STRF), which is shown in our human cortical data to yield improved predictions. We also use DCA to reconstruct speech from cortical activity, wherein dynamical dictionaries for electrocortical data are derived, with application to brain-computer interfaces (BCI).

5aPP19. Selective and divided attention: Spatial orienting in a semantic classification task. Daniel McCloy (Inst. for Learning and Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98115-7988, drmccloy@uw.edu) and Adrian KC Lee (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Preliminary data from our lab using non-lexical stimuli (alphabet letters) suggests that auditory spatial attention may act as a filter with some spatial roll-off (seen in patterns of false alarms to targets in unattended spatial

streams). The current project extends that research, examining the interaction between auditory spatial attention and lexical activation. In a semantic classification task, listeners respond to words of a target class only when they occur in designated spatial streams. Streams are temporally interleaved (in random spatial order) to minimize energetic masking. Given that tone-complex experiments suggest a short time course for exogenous spatial reorientations (at least 150 ms but less than 450 ms) [Roberts *et al.*, J. Exp. Psychol. Human **35**, 1178–1191 (2009)] when compared to lexical activation (~400 ms) [Pylkkänen *et al.*, Brain Lang **81**, 666–678 (2002)], we predict that the additional processing required for semantic classification could paradoxically reduce false alarms in spatial locations proximal to the target stream(s), by delaying the response decision beyond the temporal scope of the exogenous orienting response. We discuss findings in relation to models of exogenous/endogenous attention and lexical processing.

5aPP20. The role of syntax in maintaining the integrity of streams of speech. Gerald Kidd, Christine Mason (Speech, Lang. & Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA, gkidd@bu.edu), and Virginia Best (National Acoust. Labs., Australian Hearing Hub, Macquarie Univ., NSW, Australia)

This study examined the ability of listeners to utilize syntactic structure to extract a target stream of speech from among competing sounds. Target talkers were identified by voice or location, which was held constant throughout an utterance, and were constructed to have either correct syntax or random word order. Both voice and location provided reliable cues for identifying target speech even when other features varied unpredictably. The target sentences were masked either by predominantly energetic maskers (noise) or by highly uncertain informational maskers (similar speech in random word order). When the maskers were noise bursts, target sentence syntax had relatively minor effects on identification performance. However, when the maskers were other utterances, target sentence syntax resulted in significantly better speech identification performance. In addition, conformance to correct syntax alone was sufficient to accurately identify the target speech. The results were interpreted as support for the idea that the predictability of the elements comprising streams of speech, as manifested by syntactic structure, is an important factor in binding words together into coherent streams. Furthermore, these findings suggest that predictability is particularly important for maintaining the coherence of an auditory stream over time under conditions high in informational masking.

5aPP21. Effect of frequency variation and covariation on auditory streaming of tone sequences. An-Chieh Chang and Robert Lutfi (Dept. of Commun. Sci. and Disord., Univ. of Wisconsin - Madison,Madison, WI, achang5@wisc.edu)

Recent results from our lab show the masking of one tone sequence by another to be strongly related to the information divergence of sequences, a measure of statistical separation of signals [Gilbertson *et al.*, POMA **19**, 050028 (2013)]. The present study was undertaken to determine if the same relation holds for the auditory streaming of tone sequences. An adaptive procedure was used to measure thresholds for streaming of ABA_{ABA} tone sequences wherein the frequencies of the A and B tones varied independently of one another ($r = 0$) or covaried within the sequence ($r = 1$). The procedure adapted on the difference Δ in the mean frequencies of A and B tones (normally distributed in cents) with the mean frequency of A tones fixed at 1000 Hz. For most listeners, Δ increased monotonically with increases in the variance of the tone frequencies (σ = 0-800 cents), but did not differ significantly for $r = 0$ and $r = 1$. For other listeners, Δ was a nonmonotonic function of variance and differed for $r = 0$ and $r = 1$. The results fail to support a strong relation between auditory streaming and the information divergence of tone sequences.

5aPP22. Simultaneous recording of brain responses indicating sensation and perception of changes in interaural phase differences. Bernhard Ross (Rotman Res. Inst., Baycrest Ctr., 3560 Bathurst St., Toronto, ON M6A 2E1, Canada, bross@research.baycrest.org) and Takako Fujioka (Ctr. for Comput. Res. in Music and Acoust., Dept. of Music, Stanford Univ., Stanford, CA)

Changing the interaural phase difference (IPD) between binaurally presented tones induces the sensation of a change of the sound source in space and elicits auditory brain responses specific for sound localization. We recorded neuromagnetic responses to IPD changes in young, middle-aged, and older listeners at various tonal frequencies. Young listeners showed brain responses below 1500 Hz according to the behavioral findings of using IPD for sound localization at low frequencies only. The upper limit for IPD detection decreased with age, and older listeners (mean age of 71 years) could make use of IPD changes only for tonal sounds below 750 Hz. The stimuli were amplitude modulated at 40 Hz and elicited synchronized brain activity at the rhythm of the amplitude modulation. Thus, 40-Hz brain activity was recorded simultaneously with the IPD change responses. Although the amplitude modulation as continuous and specifically did not change the interaural phase relation, the 40-Hz brain response was reset at the IPD change. We interpret the 40-Hz brain responses as related to sensory binding for perception. Each change in the auditory environment requires a reset and reconfiguration of the binding network, which can be observed in the reset of 40-Hz brain oscillations. Recording simultaneously brain responses to sensation and perception of IPD changes gives insight into the temporal dynamics of binaural auditory processing.

5aPP23. Maximum acceptable vibrato excursion as a function of vibrato rate in musicians and non-musicians. Marianna Vatti (Eriksholm Res. Ctr., Oticon A/S, Rørtangvej 20, Snekkersten 3070, Denmark, mav@eriksholm.com), Sébastien Santurette (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark), Niels H. Pontoppidan (Eriksholm Res. Ctr., Oticon A/S, Snekkersten, Denmark), and Torsten Dau (Ctr. for Appl. Hearing Res., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

Human vibrato is mainly characterized by two parameters: vibrato extent and vibrato rate. These parameters have been found to exhibit an interaction both in physical recordings of singers' voices and in listener's preference ratings. This study was concerned with the way in which the maximum acceptable vibrato excursion varies as a function of vibrato rate in normal-hearing (NH) musicians and non-musicians. Eight NH musicians and six non-musicians adjusted the maximum vibrato excursion of a synthesized vowel for vibrato rates between 3 and 8 Hz. Individual thresholds varied across vibrato rate and, in most listeners, exhibited a peak at medium vibrato rates (5–7 Hz). Large across-subject variability was observed, and no significant effect of musical experience was found. Overall, most listeners were not solely sensitive to the vibrato excursion and there was a listener-dependent rate for which larger vibrato excursions were favored. The observed interaction between maximum excursion thresholds and vibrato rate may be due to the listeners' judgments relying on cues provided by the rate of frequency changes (RFC) rather than excursion per se. Further studies are needed to evaluate the contribution of the RFC to vibrato perception and the possible effects of age and hearing impairment.

Session 5aSCa

Speech Communication: Predictive Processing

Dan Silverman, Chair
Dept. of Linguistics, San Jose Univ., One Washington Square, San Jose, CA 95192-0093

Contributed Papers

8:30

5aSCa1. Predictive processing during discourse comprehension. Marisa Casillas (Lang. and Cognition, Max Planck Inst. for PsychoLinguist, Postbus 310, Nijmegen 6500 AH, Netherlands, Marisa.Casillas@mpi.nl) and Michael C. Frank (Linguist, Stanford Univ., Stanford, CA)

We investigate children's online predictive processing as it occurs naturally, in conversation. We showed 129 children (1;0–7;0) short videos of improvised conversation between puppets, controlling for available linguistic information through phonetic manipulation: normal, prosody only (low-pass filtered), lexical only (rhythm controlled and pitch flattened), and none (multi-talker babble). We tracked their eye movements during the videos, measuring their anticipatory looks to upcoming speakers at points of turn switch (e.g., after a question and before an answer). Even one- and two-year-old children made accurate and spontaneous predictions about when a turn-switch would occur: they gazed at the upcoming speaker before they heard a response begin. By age three, children distinguished between different types of response-eliciting speech acts, looking faster to question- than non-question responses—but only when all linguistic information was available. By age seven, children's gaze behavior also distinguished between rising and non-rising turns in the prosody only condition. These predictive skills rely on both lexical and prosodic information together, and are not tied to either type of information alone. We suggest that children integrate prosodic, lexical, and visual information to effectively predict upcoming linguistic material in conversation.

8:45

5aSCa2. Effects of emotional prosody on word recognition. Seung Kyung Kim and Meghan Sumner (Stanford Univ., 450 Serra Mall, Stanford, CA 94305, skim2@stanford.edu)

Phonetic variation in speech informs listeners not only about the linguistic message but also about talkers (e.g., gender, age, and emotion). In most episodic theories of speech perception, this indexical variation is accommodated via an acoustically-detailed exemplar lexicon. This view assumes lexical and indexical information are coupled, but speakers use acoustic patterns productively to convey information independent of the words they utter. We investigated the effects of emotional prosody on word recognition to test whether indexical information affects word recognition independent of lexical information. First, we compared the recognition of emotion word targets (UPSET) preceded by semantically unrelated words spoken with emotionally related or unrelated prosody (pineapple$_{\text{[AngryVoice]}}$ or pineapple$_{\text{[NeutralVoice]}}$). Second, we investigated the effects of emotional prosody on semantically-related target words (pineapple$_{\text{[AngryVoice]}}$ or pineapple$_{\text{[NeutralVoice]}}$—FRUIT). Recognition of both emotionally related and semantically related targets was facilitated by prime words spoken with angry prosody. These data suggest that indexical variation in speech influences word recognition beyond acoustically-detailed lexical representations. We suggest listeners simultaneously process acoustic variation for indexical and lexical meaning and argue that emotional prosody activates emotion features and categories, independent of lexical access.

9:00

5aSCa3. Phonological confusions in verbal working memory. Marc Ettlinger, E. W. Yund, Timothy J. Herron, and David L. Woods (Res. Service, Dept. of Veterans Affairs, 151/MTZ, 40 Muir Rd., Martinez, CA 94553, ettlinger@gmail.com)

Previous research has shown that phonological factors impact verbal working memory (VWM) including worse memory for phonologically similar items, for phonologically longer items and for items with low-frequency phonemes and phonotactics. These effects, and others, suggest that the substrate of VWM is phonological in nature. However, if VWM is phonological, we should expect another effect: that errors made in recall should reflect phonological principles. In the present study, we examine the errors in a verbal working memory task with stimuli that include all possible CVC words and pseudo-words. This allows not only for a corroboration of previous work on consonant transpositions (i.e., spoonerisms) in memory, but also permits an examination of the newly discovered phenomenon of substitution errors in memory. The results show that the substitution errors in verbal working memory reflect the consonant confusion errors found in speech perception, with a number of interesting exceptions. Not only do these findings introduce a novel effect of the phonological nature of WM, they also bear on the question of whether VWM is articulatory or perceptual in nature, suggesting that VWM that is based on a synthesis of both the perceptual and production systems.

9:15

5aSCa4. Information-bearing acoustic change outperforms duration in predicting sentence intelligibility in normal and simulated electric hearing. Christian Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@gmail.com)

Recent research has demonstrated a strong relationship between information-bearing acoustic changes in the speech signal and speech intelligibility. The availability of information-bearing acoustic changes robustly predicts intelligibility of full-spectrum (Stilp and Kluender 2010 *PNAS*) and noise-vocoded sentences amidst noise interruption (Stilp *et al.,* 2013 *J. Acoust. Soc. Am.*). Other research reports that duration of preserved signal also predicts intelligibility of noise-interrupted speech. These factors have only ever been investigated independently, obscuring whether one better explains speech perception. The present experiments manipulated both factors to answer this question. A broad range of sentence durations with high or low information-bearing acoustic changes were replaced by speech-shaped noise in noise-vocoded and full-spectrum sentences. Sentence intelligibility worsened with increasing noise replacement, but in both experiments, information-bearing acoustic change was a statistically superior predictor of performance. Perception relied more heavily on information-bearing acoustic changes in poorer listening conditions (in spectrally degraded sentences and amidst increasing noise replacement). Highly linear relationships between measures of information and performance suggest that exploiting information-bearing acoustic change is a shared principle underlying speech perception in acoustic and simulated electric hearing. Results demonstrate the explanatory power of information-theoretic approaches for speech perception.

9:30

5aSCa5. The impact of spectral resolution on listening effort revealed by pupil dilation. Matthew Winn (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Rm. 565, Madison, WI 53705, mwinn83@gmail.com) and Jan R. Edwards (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI)

Poor spectral resolution is a consequence of cochlear hearing loss and remains arguably the primarily limiting factor in success with a cochlear implant. In addition to showing reduced success on word recognition compared to their normal-hearing peers, listeners with hearing impairment also are reported to exert greater effort in everyday listening, leading to difficulties at the workplace and in social settings. Pupil dilation is an index of cognitive effort in various tasks, including speech perception. In this study, spectral resolution was explicitly controlled for in listeners with normal hearing using a noise vocoder with variable number of processing channels. Pupil dilation during a sentence listening and repetition task revealed a systematic relationship between spectral resolution and listening effort; as resolution grew poorer, effort increased. Significant changes in listening effort belie the notion of "ceiling" performance in degraded conditions; listeners are able to achieve success in the face of signal degradation at least partly on behalf of extra effort required to listen. We provide a model by which interventions for clinical populations (e.g., processing strategies) can be evaluated on the basis of listening effort, beyond the conventional techniques of word and sentence recognition accuracy.

9:45–10:00 General Discussion

FRIDAY MORNING, 6 DECEMBER 2013 — PLAZA B, 10:15 A.M. TO 11:45 A.M.

Session 5aSCb

Speech Communication: Neurophonetics

Marc Ettlinger, Chair
Research Service, Dept. of Veterans Affairs, 151/MTZ, 40 Muir Rd., Martinez, CA 94553

Contributed Papers

10:15

5aSCb1. Neural evidence for shared phonetic, phonological, and lexical processing of words and pseudowords. Emily Cibelli (Linguist., Univ. of California, Berkeley, 1890 Arch St., Apt. 302, Berkeley, CA 94709, ecibelli@berkeley.edu), Matthew Leonard, and Edward Chang (Depts. of Neurological Surgery and Physiol. and Ctr. for Integrative Neurosci., Univ. of California, San Francisco, San Francisco, CA)

This study uses electrocorticography (ECoG) to investigate word and pseudoword auditory processing. ECoG data are recorded from intracranial electrodes with high spatial and temporal resolution. This methodology contributes novel data to the debate over whether words and pseudowords are processed using shared streams, or whether pseudowords rely on separate sub-lexical routes. Data from left temporal lobe electrodes was recorded from two patients in a listen-and-repeat task with real words (e.g., "minority") and pseudowords (e.g., [təmiɹɪnai]). For each electrode showing a word/pseudoword difference, regression models were fit to capture the time-varying effects of lexicality, cohort size (how many lexical items matched the current phonetic input), and cohort frequency. Preliminary results show that lexical factors had predictive power in mid- and anterior temporal electrodes. Activity peaked early in posterior electrodes and propagated forward to anterior sites. Average activity was stronger for pseudowords than words. A positive relationship was found between cohort frequency and activity; the direction of the effect varied for cohort size. The data is consistent with a shared streams account: along the temporal lobe, words and pseudowords share processing in acoustic, phonetic, phonological, and lexical regions, with access to stored lexical/cohort information.

10:30

5aSCb2. Neural connectivity of voice control using structural equation modeling. Sabina Flagmeier, Kimberly L. Ray, Amy L. Parkinson (UT Health Sci. Ctr. San Antonio, 7703 Floyd Curl Dr., San Antonio, TX 78251, gonzalessm@uthscsa.edu), Angela R. Laird (Phys., Florida Int. Univ., Miami, FL), Victoria Folks (UT Health Sci. Ctr. San Antonio, San Antonio, TX), Charles Larson (Commun. Sci. and Disord., Northwestern Univ., San Antonio, IL), and Donald A. Robin (UT Health Sci. Ctr. San Antonio, San Antonio, TX)

Introduction: This study aims to model connectivity of neural regions involved in voice control. Here, we used structural equation modeling on a published dataset that employed the pitch shift paradigm. We hypothesized that our models would confirm differences in connectivity related to superior temporal gyrus during error processing of vocalization. Methods: We extracted time course data of eight regions included from 10 healthy subjects. A detailed description of subjects, MRI scanning procedures, imaging acquisition and data analysis can be found in Parkinson *et al.* 2012. Effective connectivity of regions activated during shift and no-shift paradigms was assessed using structural equation modeling techniques (AMOS version 19.0, SPSS, IBM). Results Consistent with our hypothesis, STG appears to play a crucial role in vocalization and error processing, showing increased participation of the right hemisphere during the shift condition than the no shift condition. Furthermore, left inferior frontal gyrus displays significant contribution to the modulation of vocal control through connections with PMC that change in response to the shift condition. Conclusions: Results indicated changes in connectivity of the voice network related to error detection and correction. Our models indicate hemispheric sensitivity to different elements of the auditory feedback and highlight the importance of examining network connectivity.

10:45

5aSCb3. A right-lateralized cortical network drives error correction to voice pitch feedback perturbation. Naomi Kort (BioEng., Univ. of California, San Francisco, 513 Parnassus Ave., S362, San Francisco, CA 94143-0628, naomi.kort@ucsf.edu), Srikantan S. Nagarajan (Radiology, Univ. of California, San Francisco, San Francisco, CA), and John F. Houde (Otolaryngol., Univ. of California, San Francisco, San Francisco, CA)

One of the most intriguing discrepancies in speech neuroscience arises from data on laterality: lesion studies have provided overwhelming evidence for a left-dominant model of speech production, yet neuroimaging studies consistently show bilateral neural activity in speech related tasks. Recently, a model has been proposed to resolve this discrepancy. This model suggests that the left hemisphere generates feed-forward production of speech, while the right hemisphere, specifically right frontal regions, monitors and responds to feedback for ongoing control. Using real-time pitch-altered auditory feedback and magnetoencephalography, we demonstrate that the right hemisphere subserves feedback control of pitch production. During ongoing phonation, speakers respond rapidly to pitch shifts of their auditory feedback, altering their pitch production to oppose the applied pitch shift. Immediately following the onset of the pitch shift, bilateral sensorimotor cortex shows an increase in high gamma power. Yet, within 100 ms, the responses in the left hemisphere decrease and are limited to one region of left posterior temporal cortex while the high gamma power in the right hemisphere increases in premotor cortex, ventral supramarginal gyrus, inferior and middle frontal gyrus. These findings provide evidence for key roles for right premotor and right SMG in making small, rapid compensations to feedback errors.

11:00

5aSCb4. Human superior temporal gyrus encoding of speech sequence probabilities. Matthew K. Leonard, Kristofer Bouchard, and Edward F. Chang (Neurological Surgery, UCSF, 675 Nelson Rising Ln., Rm. 510, San Francisco, CA 94158, leonardm@neurosurg.ucsf.edu)

Spoken word representations are hypothesized to be built from smaller segments of the speech signal, including phonemes and acoustic features. The language-level statistics of sound sequences ("phonotactics") are speculated to play a role in integrating sub-lexical representations into words in the human brain. In four neurosurgical patients, we recorded electrocorticographic (ECoG) neural activity directly from the brain surface while they listened to spoken real and pseudo words with varying transition probabilities (TPs) between the consonants and vowels (Cs and Vs) in a set of CVC stimuli. Electrodes over left superior temporal gyrus (STG) were sensitive to TPs in a way that suggested dynamic, near real-time tracking of the speech input. TP effects were seen independently from activity explained by acoustic variability as measured by each electrode's spectrotemporal receptive field (STRF). Furthermore, population-level analyses of STG electrodes demonstrated that TP effects were different for real vs pseudo words. These results support the hypothesis that lifelong exposure to phonetic sequences shapes the organization and synaptic weights of neural networks that process sounds in a given language, and that phonotactic information is used dynamically to integrate sub-lexical speech segments toward lexical representations.

11:15

5aSCb5. Cortical processing of audiovisual speech perception in infancy and adulthood. Yang Zhang (Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, zhang470@umn.edu), Bing Cheng (Xi'an Jiaotong Univ., Minneapolis, Minnesota), Tess Koerner, Christine Cao, Edward Carney (Univ. of Minnesota, Minneapolis, MN), and Yue Wang (Simon Fraser Univ., Burnaby, BC, Canada)

The ability to detect auditory-visual correspondence in speech is an early hallmark of typical language development. Infants are able to detect audiovisual mismatches for spoken vowels such as /a/ and /i/ as early as 4 months of age. While adult event-related potential (ERP) data have shown an N300 associated with the detection of audiovisual incongruency in speech, it remains unclear whether similar responses can be elicited in infants. The present study collected ERP data in congruent and incongruent audiovisual presentation conditions for /a/ and /i/ from 21 typically developing infants (6~11 month of age) and 12 normal adults (18~45 years). The adult data replicated the N300 in the parietal electrode sites for detecting audiovisual incongruency in speech, and minimum norm estimation (MNE) showed the primary neural generator in the left superior temporal cortex for the N300. Unlike the adults, the infants showed a later N400 response in the centro-frontal electrode sites, and scalp topography as well as MNE results indicated bilateral activation in the temporal cortex with right-hemisphere dominance. Together, these data indicate important developmental changes in the timing and hemispheric laterality patterns for detecting audiovisual correspondence in speech.

11:30–11:45 General Discussion

FRIDAY MORNING, 6 DECEMBER 2013 PLAZA A, 8:00 A.M. TO 12:00 NOON

Session 5aSCc

Speech Communication: Speech Analysis (Poster Session)

Robert Podesua, Chair
Stanford Univ., Stanford, CA 94305-2150

Contributed Papers

5aSCc1. The role of memory and representations in statistical learning. Alexis Black (Linguist, Univ. of Br. Columbia, Totem Field Studios 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, akblack2g@gmail.com)

Numerous studies have examined learners' ability to track auditory statistical cues (e.g., Saffran, Aslin, and Newport, 1996). It remains unknown, however, how learners manage this feat. This concern is non-trivial: computation of the transitional probabilities in a traditional statistical learning task would involve access to (at least) hundreds of memory traces that have been accumulated over a mere two minutes. The present experiments aim to elucidate the mechanisms underlying statistical learning. Adult participants are exposed to a 2-min continuous speech stream, composed of native phonetic units (study 1), semi-native phonetic units (study 2), or non-

native phonetic units (study 3). Participants' memories for words are then tested through forced-choice comparisons of words, part-words, and phantom words. In study 1, participants successfully segmented the native phonetic speech stream; however, they showed no preference for part-words or phantom words. Furthermore, asymmetries in performance by syllable position (onset, medial, coda) suggest that memory for the segmented words may be more specified/stable in the medial and coda positions. Preliminary results from study 3 (non-native) suggest that participants fail to segment the stream of less familiar sounds. Taken together, these results suggest that statistical learning proceeds via a "chunking"-type mechanism (e.g., Perruchet and Vinter, 1998).

5aSCc2. Using tactile aids to provide low frequency information for cochlear implant users. Shuai Wang, Xuan Zhong, Michael F. Dorman, William A. Yost, and Julie M. Liss (Dept. of Speech and Hearing Sci., Arizona State Univ., PO Box 870102, Tempe, AZ 85287, swang102@asu.edu)

Cochlear implant (CI) users have shown benefit from residual low-frequency hearing in the contra-lateral ear (Dorman and Gifford, 2010). One source of this benefit is the enhancement of cues important for identifying word boundaries (Spitzer *et al*., 2009). However, there are a large number of CI users who do not have residual hearing, but who could presumably benefit from cues available in low-frequency information. Because the frequency sensitivity of human haptic sensation is similar to that of human acoustic hearing in low frequencies, we examined the ability of tactile aids to convey low-frequency cues. Using experimental phrases designed to have low inter-word predictability, and balanced for syllabic stress (trochaic/iambic), 5 CI users and 10 normal hearing participants (simulation) provided transcriptions that were scored for percent words-correct and for errors in word segmentation (lexical boundary errors, LBE). A 350 Hz sinusoid carrier modulated with overall envelope of corresponding acoustic signal drove two bone-anchored hearing aids (BAHA), which participants held while listening. Results showed a small but significant improvement on percent words correct with tactile aid, and fewer word segmentation errors. These findings support the benefit of tactile information in the perceptual task of lexical segmentation.

5aSCc3. The effect of aging on auditory processing: Temporal resolution and informational masking. Won So and Su-Hyun Jin (Commun. Sci. and Disord., Univ. of Texas at Austin, 1 University Station, A1100, Austin, TX 78712, shjin@utexas.edu)

This study examined age-related changes in temporal resolution and speech perception in noise. Older listeners tend to exhibit more difficulty listening in noise, especially, understanding speech in complex noise, such as temporally modulating noise. The current study examined younger and older listeners for their understanding of speech in spectrally remote modulating noise. When the spectrum of noise is distant from that of speech, the effect of energetic masking would be minimized, leading us to measure the effect of informational masking on speech perception. We hypothesized that older listeners may show a significant amount of informational masking even when the noise spectrum is distant from the speech spectrum due to greater central interference of the noise compared to younger listeners. We also measured pure tone glide detection in steady and gated noise (Nelson *et al*., 2011). When the pure tone frequency changes from low to high (or high to low), which is similar to spectral change in speech, older listeners might have more difficulty detecting glides in modulating noise than younger listeners because they would not be able to detect spectral changes available in the brief dips in the modulating noise.

5aSCc4. Evaluation of percentage hearing loss formulae. Colette Vossler-Welch (Commun. Disord., Utah State Univ., 30930 Peterson Rd., Philomath, Oregon 97370, cbvossler@hotmail.com) and Ron J. Leavitt (Audiol., Corvallis Hearing Ctr., Independance, OR)

Across the United States, those who lose their hearing in the workplace or while in the military are compensated by a formula that attempts to assign a percent of disability value to the hearing loss. Since current medical practice strives toward high-quality scientific evidence to guide the practitioner one might assume the calculation for percent hearing disability would be evidence-based and utilize level-one, scientifically validated computations. To the contrary, we have not found a strong scientific foundation for these percent hearing loss calculations in refereed scientific journals. Results from this study will show that current percent disability computations do not correlate well with the difficulties patients report in their everyday listening environments. These findings suggest a new formula is needed to compute percent disability that accurately portrays the communication difficulties of people with hearing loss.

5aSCc5. Lexical tone and consonant perception in subtypes of Schizophrenia. Feng-Ming Tsao (Psych., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 106, Taiwan, tsaosph@mail2000.com.tw), Shih-Kuang Chiang (Counseling and Clinical Psych., National Dong Hwa Univ., Hualien, Taiwan), and Huei-Mei Liu (Special Education, National Taiwan Normal Univ., Taipei, Taiwan)

Auditory hallucination is one of diagnostic criteria of schizophrenia and might negatively affect speech perception. Among subtypes of schizophrenia, the persistent delusion/hallucination (PDH) group persistently shows auditory hallucinations after 6 months of admission. This study aims to examine whether hallucinations affect lexical tone and consonant perception in Mandarin-speaking adults. Two groups of adults with chronic schizophrenia, PDH ($n = 15$, mean age $= 44$ yr) and non-hallucination group ($n = 17$, mean age $= 39$ yr), and age-matched control group ($n = 16$, mean age $= 36$ yr) in Taiwan were tested. For lexical tone perception, results showed that adults with schizophrenia were less accurate than typical controls on discriminating the lexical tones, and the PDH group performed poor than patients without hallucination. The lexical tone accuracy negatively correlates with the severity of schizophrenic symptoms ($r = -0.559$, $p < 0.001$, measured with Positive and Negative Syndrome Scale for schizophrenia). For consonant perception, patient groups showed poor perceptual organizations for affricates than control group. Moreover, the perceptual organization of PDH group is more distorted than non-hallucination group. In brief, adults with chronic schizophrenia exhibit speech perception deficits, and these deficits might be the result of a distorted perceptual organization.

5aSCc6. The effect of working memory capacity on sequencing errors in child speech. Wook Kyung Choe and Melissa A. Redford (Dept. of Linguist., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403-1290, wchoe1@uoregon.edu)

The current study investigated the effect of working memory on the distribution and other characteristics of sequencing errors in school-aged children's production of sentence-length tongue twisters. Our goal was to understand the relationship between memory and speech planning in child speech. Working memory was assessed using subtests from the CTOPP (Wagner *et al*., 1999). Errors were elicited by asking 33 children (6-to-9-year-olds) to read different tongue twisters multiple times. Anticipatory and perseveratory errors were identified and categorized; strong and weak prosodic boundaries were also identified. Results showed that the children with larger working memory capacity produced significantly shorter prosodic phrases than those with smaller working memory capacity. Otherwise, the distribution and other characteristics of errors in the former group were more adult-like than those of the latter group: more errors toward the end of prosodic phrases (Choe and Redford, 2012); lower error rates (Wijnen, 1992); and more anticipatory than perseveratory errors (Vousden and Maylor, 2006). We suggest that these results are consistent with more structured speech plans in children with larger working memory capacity than in those with smaller capacity. [This research was supported by NICHD.]

5aSCc7. Learning words from multiple talkers helps children's production but not perception. Andrea K. Davis (Linguist, Univ. of Arizona, 1076 Palomino Rd., Cloverdale, California 95425, davisak@email.arizona.edu) and LouAnn Gerken (Linguist., Univ. of Arizona, Tucson, AZ)

Phonetic variation between speakers promotes generalization when a listener is learning new speech sounds or new word forms (Lively *et al*., 1993;

Richtsmeier *et al*., 2009; Rost and McMurray, 2009, 2010). In the latter two studies, infant learners were better able to discriminate newly learned words produced by new talkers when trained with multiple talkers. But is variation always helpful for generalization? A variety of factors may influence whether variation is beneficial, including the amount of prior experience with the language, or whether the test is on perception vs production. A study with pre-schoolers addresses these potential influences on whether variation is helpful for learning word forms. Children aged 2.5–5 learned four new words, from either multiple talkers or a single talker. They were then asked to detect a puppet's mispronunciations, and then to produce the new words. Results suggest that for older as well as younger pre-schoolers, learning with variation does not help with detecting mispronunciations of the word. However, results of Richtsmeier *et al*. are replicated, with children producing words more accurately when they learned the words from multiple talkers. This suggests that speakers use different representations for perception and production, at least when a word is newly learned.

5aSCc8. Graph alignment and cross-modal learning during early infancy. Andrew R. Plummer (Ohio State Univ., 1712 Neil Ave., Columbus, OH 43210, plummer@ling.ohio-state.edu)

Results of decades of research on vowels support the conclusion that perception and production of language-specific vowel categories cannot be based on invariant targets that are represented directly in either the auditory domain or the articulatory (sensorimotor) domain. This raises a number of questions about how an infant can acquire the cognitive representations relevant for learning the vowels of the ambient language. Some models of the acquisition process assume a fixed auditory transform to normalize for talker vocal tract size (e.g., Callan *et al*., 2000), ignoring evidence that normalization must be culture-specific (e.g., Johnson, 2005). Others assume that learning can be based on statistical regularities solely within the auditory domain (e.g., Assmann and Nearey, 2008), ignoring evidence that articulatory experience also shapes vowel category learning (e.g., Kamen and Watson, 1991). This paper outlines an alternative approach that models cross-modal learning. The approach aligns graph structures, called "manifolds," which organize sensory information in the auditory and in the articulatory domain. Graph alignment is guided by perceptual targets that are internalized in early infancy through social/vocal interaction with caregivers, so that vowel categories can be identified with the abstractions that mediate between the two domains in the alignment process.

5aSCc9. Production and perception of tones by English speaking children. Irina A. Shport (Dept. of Linguist, Univ. of Oregon, 260-G Allen Hall, Baton Rouge, Louisiana, ishport@lsu.edu), Melissa A. Redford (Dept. of Linguist, Univ. of Oregon, Eugene, OR), and Bharath Chandrasekaran (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

Production and perception may not be correlated in adult learners of tonal languages (Bent, 2005). We examined whether these abilities might however be correlated in children. Thirty-one children (age 7;1–9;0) and 20 adults participated in tone learning experiment. They repeated 8 "Martian" words, varying in tone patterns (high, rising, falling-rising, falling) and in tone location (first syllable, second syllable), 4 times in random order over each training block (3). Production accuracy was assessed using discriminant analysis. Mean F0, excursion size, final velocity, and duration were the predictor variables. Tone discrimination was measured in an AX task based on low-pass versions of the same "words" used in training. Children's working memory capacity, a known predictor of first language acquisition (Gathercole and Baddeley, 1993), was also measured. The results revealed no correlation between production and perception in adults. In contrast, production accuracy in the first training block, perceptual discrimination (d′), and working memory combined to predict production accuracy in the final training block ($R^2 = 0.43$) in children. The results suggest that sensitivity to pitch variation and working memory may influence second language tone learning in children. [Work support by NICHD.]

5aSCc10. Effects of musical rhythm training on infants' neural processing of temporal information in music and speech. Tian Zhao and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu)

Investigations of musical training provide a way to study neural plasticity within the domain of music, as well as to study transfer effects to other domains (speech). Previous research has reported anatomical and functional differences between musically trained and non-trained individuals. However, these studies have not addressed several issues, including (1) nature vs nurture, (2) timing of musical training, and (3) the temporal aspect (e.g., rhythm) of musical training rather than the frequency aspect (e.g., pitch). The current study aims to examine of the causal effect of musical training on the sensitivity to temporal information in both music and speech sounds in infancy. In the study, 9-month-old infants were randomly assigned to a 12-session musical training condition vs a control condition (mirroring the design of Kuhl *et al*., 2003). During training, infants were exposed to uncommon metrical structure in music through social, multimodal, and structured activities while infants played freely in the control condition. After training, infants' sensitivities to occasional violations in temporal structure were examined, both in music and speech. A traditional oddball paradigm was used and infants' neural activities were recorded by MEG. Data from pilot participants will be discussed. [Research supported by I-LABS' Developing Mind Project.]

5aSCc11. Contextual influences on speech perception in developmental populations. Rachel M. Theodore, Jean Campbell, MaryKate Bisaillon, and Devin Roscillo (Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

A major goal of speech perception research has been to describe how listeners recognize individual consonants and vowels from the speech stream given rampant acoustic-phonetic variability in their instantiation. Findings in healthy adults indicate that listeners achieve perceptual stability, at least in part, by dynamically adjusting phonetic boundaries to accommodate contextual influences in speech production. The current work examines the developmental trajectory of such functional plasticity in typically developing children. Across two experiments, we examined the influence of speaking rate and place of articulation on stop consonant identification. Stimuli consisted of three voice-onset-time continua: "goal" to "coal" at a fast speaking rate, "goal" to "coal" at a slow speaking rate, and "bowl" to "pole" at a slow speaking rate. The results showed that 8-10-year-old children are sensitive to how these contextual influences pattern in speech production. Specifically, the identification responses indicated that the voicing boundary was located at longer VOTs for the slow compared to the fast speaking rate continuum and for the velar compared to the labial continuum. These findings suggest that perceptual sensitivity to contextual influences in speech production emerges early in development, illustrating a critical role for functional plasticity toward the healthy end-state system.

5aSCc12. Assessing whether loud speech affects vowel formant values in toddlers. Laura L. Koenig and Jonathan Preston (n/a, Haskins Labs., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu)

Extensive token-to-token variability is a widely-noted characteristic of child speech. This variability likely arises from several sources. In a recent analysis of vowel formants in the speech of toddlers, we observed that some children produced extreme ranges of F1 for the same target vowel, and that some high values of F1 were associated with a loud voice. Here, we undertake a systematic analysis of vowel formants as a function of perceived loud voice. Data were obtained from children 2–3 years of age producing many repetitions of the words "baby," "ball," "boy," "bubble," "moo," and "pooh." Since mouth-to-microphone distance varied, an acoustic measure of loudness was not possible. However, speaking level can affect voice quality and possibly other production features, and might be reliably perceptible nevertheless. All word productions suitable for acoustic analysis will be auditorily assessed by naïve listeners as having "regular voice" or "loud voice." We will then determine whether productions judged as loud vary systematically in their formant frequencies. If so, it would suggest that researchers studying young child speech should attempt to limit extreme variations in vocal loudness, particularly if they are interested in acoustic measures of variability.

5aSCc13. Visual and sensori-motor influences on speech perception in infancy. D. Kyle Danielson, Alison J. Greuel, and Janet F. Werker (Psych., Univ. of Br. Columbia, 2136 West Mall, Vancouver, BC V6T 1Z4, Canada, kdanielson@psych.ubc.ca)

Speech perception is multisensory and is comprised not only of auditory information, but also of visual (Burnham and Dodd, 2004; Kuhl and Meltzoff, 1984; Patterson and Werker, 2003) and proprioceptive motor information (Yeung and Werker, 2013). Building on previous studies examining perceptual attunement in young infants (Werker and Tees, 1984, *inter alia*), this set of experiments examines the role that vision and motor proprioception play during the perception of auditory speech sounds. In the first experiment, the developmental trajectory of perceptual attunement in English-learning infants is again explored using the dental-retroflex contrast of Hindi. We replicate the finding that English-learning 6-month-olds are able to discriminate auditory-only dental and retroflex stimuli, while English-learning 10-month-olds are not. In the second experiment, looking time measurements are used to explore the possibility that the addition of dynamic visual information acts as a perceptual anchor, now permitting discrimination of this same non-native contrast by 10-month-old infants. Finally, in the third experiment, we investigate the role of a temporary motor manipulation, designed to prevent relevant movement of the tongue in 6- and 10-month-old English infants during perception of the non-native Hindi contrast, to determine the effect of proprioceptive, sensori-motor mechanisms in auditory speech perception across development.

5aSCc14. Modeling the perception of speaker age and sex in children's voices. Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, MS GR 41, Box 830688, Richardson, TX 75075, assmann@utdallas.edu), Santiago Barreda, and Terrance M. Nearey (Linguist, Univ. of AB, Edmonton, AB, Canada)

At previous meetings, we presented data on the perception of speaker sex and age in children's voices. The stimuli common to these experiments were /hVd/ syllables in isolation and sentence context. Here we present the results of a modeling study in which acoustic measurements of the /hVd/ syllables were used to predict listener judgments of age and sex. Variables were selected based on preliminary analyses and suggestions from the literature: (1) duration; (2) average fundamental frequency; (3) geometric mean of F1 F2 F3; (4) magnitude difference between harmonics 1 and 2; (5) magnitude difference between harmonic 1 and F3 peak; (6) Cepstral pitch prominence; (7) Harmonic-to-noise ratio. Logistic regression models were constructed to predict listeners' judgments of speaker sex, and mixed effects linear regression models for speaker age. Results confirmed the importance of F0, formant frequencies and measures related to the voicing source for both age and sex. Regression coefficients for judgments of age and sex were similar to those for veridical age and sex when regressed on the same physical measures, suggesting a near-optimal use of cues by listeners.

5aSCc15. Data-driven intonational phonology. Gopala Krishna Anumanchipalli, Alan W Black (Lang. Technologies Inst., Carnegie Mellon Univ., 5000 Forbes Ave., GHC 5705, LTI, Pittsburgh, PA 15213, gopalakr@cs.cmu.edu), and Luis C. Oliveira (INESC-ID/IST Lisboa, Instituto Superior Técnico, Lisboa, Portugal)

Intonational Phonology deals with the systematic way in which speakers effectively use pitch to add appropriate emphasis to the underlying string of words in an utterance. Two widely discussed aspects of pitch are the pitch accents and boundary events. These provide an insight into the sentence type, speaker attitude, linguistic background, and other aspects of prosodic form. The main hurdle, however, is the difficulty in getting annotations of these attributes in "real" speech. Besides being language independent, these attributes are known to be subjective and prone to high inter-annotator disagreements. Our investigations aim to automatically derive phonological aspects of intonation from large speech databases. Recurring and salient patterns in the pitch contours, observed jointly with an underlying linguistic context are automatically detected. Our computational framework unifies complementary paradigms such as the physiological Fujisaki model, Autosegmental Metrical phonology, and elegant pitch stylization, to automatically (i) discover phonologically atomic units to describe the pitch contours and (ii) build inventories of tones and long term trends appropriate for the given speech database, either large multi-speaker or single speaker databases, such as audiobooks. We successfully demonstrate the framework in expressive speech synthesis. There is also immense potential for the approach in speaker, style, and language characterization.

5aSCc16. Improving speech enhancement algorithms by incorporating visual information. Ender Tekin, James Coughlan, and Helen Simon (Smith-Kettlewell Eye Res. Inst., 2318 Fillmore St., San Francisco, CA 94115, ender@ski.org)

In speech perception, the visual information obtained by observing the speaker's face can account for up to 6 and 10 dB improvements in the presence of wide-band Gaussian and speech-babble noise, respectively. Current hearing aids and other speech enhancement devices do not utilize the visual input from the speaker's face, limiting their functionality. To alleviate this shortcoming, audio-visual speech enhancement algorithms have been developed by including video information in the audio processing. We developed an audio-visual voice activity detector (VAD) that combines audio features such as long-term spectral divergence with video features such as spatio-temporal gradients of the mouth area. The contributions of various features are learned by maximizing the mutual information between the audio and video features in an unsupervised fashion. Segmental SNR (SSNR) values were estimated to compare the benefits of audio-visual and conventional audio-only VADs. VAD outputs were utilized by an adaptive Wiener filter to estimate the noise spectrum, and enhance speech corrupted by Gaussian and speech-babble noise. The SSNR improvements were similar in low-noise conditions, but the output using the audio-visual VAD was on average 8 dB better in high-noise. This shows that video can provide complementary information when audio is very noisy, leading to significant performance improvements.

5aSCc17. Performance analysis of a matcher in a lexical access system based on landmarks and distinctive features. Jess Kenney, Jason Paller-Rzepka, Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Speech Commun. Group, Res. Lab. of Electronics, MIT, 50 Vassar St., 36-513, Cambridge, MA, sshuf@mit.edu)

Performance characteristics of a prototype matcher in a lexical access system based on landmarks and distinctive features are analyzed. A database of 16 CONV files containing spontaneous American English utterances produced by 8 female speakers is annotated with words, and phone sequences derived from the word sequences are generated using the CMU phone-based dictionary. Predicted landmark and distinctive feature sequences are then generated using context-dependent rules from the phone sequences. These labels are used to map back to a lexicon which is also represented in terms of landmarks and distinctive features. The results for using core lexicons consisting of words within a CONV file show an average match rate of about 23% using only manner-class-related landmarks, and about 93% using the distinctive feature labels. Using an expanded lexicon combining all core lexicons lowers average match rates, by about 7% using landmark labels, and by 4% using the distinctive feature labels. These results provide characteristic rates for using linguistically motivated features to match to a lexicon, for both the landmark labels and for the more detailed distinctive feature labels.

5aSCc18. Real-time speech masking using electromagnetic-wave acoustic sensors. John f. holzrichter (Lawrence Livermore Lab., 200 Hillcrest Rd., Berkeley, CA 94705, jfholz@gmail.com), Lawrence C. Ng (Lawrence Livermore Lab., Hayward, CA), and John Chang (Lawrence Livermore Lab., San Leandro, CA)

Voice activity sensors commonly measure voiced-speech-induced skin vibrations using contact microphones or related techniques. We show that micro-power EM wave sensors have advantages over acoustic techniques by directly measuring vocal-fold motions, especially during closure. This provide 0.1 ms timing accuracy (i.e., ~10 kHz bandwidth) relative to the

corresponding acoustic signal, with data arriving ~0.5 ms in advanced of the acoustic speech leaving the speaker's mouth. Preceding or following unvoiced and silent speech segments can then be well defined. These characteristics enable anti-speech waves to be generated or prior recorded waves recalled, synchronized, and broadcast with high accuracy to mask the user's real-time speech signal. A particularly useful masking process uses an acoustic voiced signal from the prior voiced speech period which is inverted, carefully timed, and rebroadcast in phase with the presently being spoken acoustic signal. This leads to real-time cancellation of a substantial fraction of the voiced acoustic energy, as well as providing timing to mask the remaining un-canceled voiced speech energy, and unvoiced speech and silence periods.

5aSCc19. Direct to reverberation ratio based two channel dereverberation for automatic speech recognition. Soonho Baek and Hong-Goo Kang (Dept. of Elec. and Electron. Eng. School of Eng., Yonsei Univ., B601 134 shinchondong seodaemoon-gu, 120-749, Seoul KS013, South Korea, bestboybsh@dsp.yonsei.ac.kr)

This paper proposes a spectral subtraction based two channel dereverberation algorithm for automatic speech recognition (ASR). By observing the fact that the accuracy of ASR system with reverberant speech highly relates to the amount of reverberant component, this paper focuses on designing a novel reverberant component estimator. Especially, the proposed algorithm utilizes the relationship between direct to reverberation ratio (DRR) and the power of reverberant components, then the estimated value is further adjusted to maximize the word accuracy of recognizer. Experimental results verify that the proposed estimator improves ASR performance in various reverberant environments.

5aSCc20. The (null) effect of spectral estimator on the estimation of spectral moments. Patrick Reidy (Dept. of Linguist., The Ohio State Univ., 24A Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, patrick.francis.reidy@gmail.com)

The spectra of English voiceless sibilants [s] and [š], when computed with traditional estimators, such as the DFT calculated over an interval multiplied with a data window, exhibit relatively large variance, which is believed to introduce error in the estimation of linguistically meaningful features, such as the first four spectral moments (centroid, variance, skewness, and kurtosis). In an effort to reduce this error, it is becoming common practice to compute such features from a multitaper spectrum (MTS)—an estimator, which asymptotically has a fraction of the DFT's variance. However, while the difference in variance has been demonstrated mathematically when the number of data samples approaches infinity, it remains an open question whether the MTS engenders more precise spectral features when estimated from the short intervals that are relevant for comparing [s] and [š]. To evaluate this issue empirically, the first four moments were estimated, with an MTS and a hamming-windowed DFT, from the middle 40-ms of 2061 word-initial tokens of [s] and [š] in utterances of English words recorded by 80 children and 20 adult native speakers. Paired *t*-tests revealed no significant difference between the two estimators for any of the spectral moments.

5aSCc21. Technique for mapping fast speech. Jesse Lawrence and Lucia da Silva (Linguist., Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, jesse.lawrence@alumni.ubc.ca)

This study investigates a novel technique for mapping normal speech to fast speech based on extracted F0 and amplitude contours. Standard methods of increasing the rate of recorded speech use mechanical or digital linear time compression, producing a speech signal that is shorter in duration. However, these methods shift all acoustic parameters in linear fashion without taking into consideration the interaction between speaking rate and stress patterns. This is sharply contrasted with the nonlinear effects observed when a speaker naturally increases speech rate. Our approach, which makes use of TANDEM-STRAIGHT (Kawahara *et al.* 2008), a speech analysis and resynthesis system, compares the F0 and amplitude contours of normal speech to the F0 and amplitude contours of naturally produced fast speech in order to derive coefficients, which are used to map novel instances of normal speech to fast speech. The resulting fast speech shows more of the nonlinear characteristics of naturally produced fast speech by attending to the interaction between speaking rate and stress patterns, and eliminates the need for post-process pitch correction inherent to time compression methods. This technique therefore represents an advance in the state of the art, with applications in research and commercial technology.

5aSCc22. Gender differences in the acoustic realization of creaky voice: Evidence from conversational data collected in Northern California. Robert J. Podesva (Linguist., Stanford Univ., Bldg. 460, Margaret Jacks Hall, Stanford, CA 94305-2150, podesva@stanford.edu) and Anita Szakay (Linguist., Queen Mary, Univ. of London, Stanford, California)

Although several sociophonetic studies report greater breathiness among female speakers, a pattern often attributed to sexual dimorphism in vocal fold physiology (Södersten and Lindestad, 1990), recent studies in North America report that young women use more creaky voice than men (Yuasa, 2011; Podesva, 2013). While these recent studies examine conversational data, they rely on auditory techniques to identify creaky phonation, leaving its acoustic realization in conversational speech largely unexplored. The present study investigates the acoustic properties of creaky voice in hour-long sociolinguistic interviews with 30 speakers (15 females, 15 males; age 18–86) from Northern California. Measures of spectral tilt were taken at the midpoint of all vowels in the corpus ($N = 362{,}429$), and data were fitted to a mixed effects linear regression model. As expected, several linguistic factors influence H1-H2 values (previous and following segment, intensity, F0, F1, vowel duration, stress, phrase position, phrase duration), alone and in interaction. With regard to social factors, H1-H2 is significantly lower for female speakers, indicating greater creak, even though the H1-H2 measure under-captures creakiness for female speakers (Simpson, 2012), and no age effect was observed. In sum, while females are generally creakier, apparent time data do not indicate that this is a recent trend.

5aSCc23. The Meet a Friend corpus of spontaneous speech: New data, initial results. Tania Henetz (Psych., Stanford Univ., Stanford, CA) and Marisa Casillas (Lang. and Cognition, Max Planck Inst. for PsychoLinguist, Postbus 310, Nijmegen 6500 AH, Netherlands, Marisa.Casillas@mpi.nl)

We introduce a new collection of 60 spontaneous speech recordings that we are making available to the wider linguistic community. We video and audio recorded sixty pairs of American English speakers as they talked freely for 20 min about four general topics (e.g., pets, food, movies). Half of the pairs came in as friends, half as strangers. The corpus contains one third each of female-female, female-male, and male-male speaker pairs. Before the recording, each participant completed a Ten Item Personality (TIPI) assessment. Afterwards, each participant gave a review of their and their partner's behavior during the conversation. Each recording is then transcribed in three passes by separate transcribers and applied to the audio recording using the Penn Phonetics Lab Forced Aligner for extended search and automated extraction abilities. We present a few initial results using these new data. For example, by extracting turn-switch gaps and comparing them to participant ratings, we find support from these naturalistic data for prior, controlled experimental work showing that inter-turn gap times relate to social evaluations of the ongoing interaction. We compare disfluency between friend- and stranger-pairs, linking these patterns to any disfluency accommodation that occurred during the interaction.

Session 5aUW

Underwater Acoustics and Acoustical Oceanography: Sediment Acoustics: Modeling, Measurement, and Inversions II

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Chair's Introduction—7:55

Invited Papers

8:00

5aUW1. High-frequency sediment acoustics over transitions from dense shell hash to mud: Repeat surveys at 7 frequencies from 150 kHz to 450 kHz. Christian de Moustier (HLS Res., Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, cpm@hlsresearch.com) and Barbara J. Kraft (HLS Res., Inc., Barrington, New Hampshire)

Seafloor acoustic backscatter data were collected with a high-frequency multibeam echo-sounder offshore Panama City Beach, FL, in May 2013, as part of the Target and Reverberation Experiment 2013 (TREX13) sponsored by the Office of Naval Research (ONR). In this context, 7 repeat surveys of a 3 km^2 area were done at individual frequencies ranging from 150 to 450 kHz, in 50 kHz increments. The regional seafloor terrain is characterized by a ridge-swale topography. Sediments in the area surveyed include mixtures of sand and mud in various proportions, augmented with shell hash whose distribution appears to be driven by bottom currents. This is inferred from maps of acoustic backscatter intensity data that show sharp boundaries (>10 dB) between dense shell hash accumulations and mud. These sediment acoustic transitions occur over a few meters and extend across the survey area, usually at the bottom of a swale. The transition pattern is consistent at all frequencies in the 7 maps of acoustic backscatter intensity (one per frequency). [Work funded by ONR Code 32OA, with sonar technical support by Teledyne-RESON.]

8:20

5aUW2. Correlations between transmission loss measurements and sediment type. Jacob George and David W. Harvey (Code 532, NAVOCEANO, 1002 Balch Blvd., Stennis Space Ctr., MS 39522, jacob.george1@navy.mil)

We report the results of a study correlating mid-frequency transmission loss (TL) measurements with properties of nearby sediment core samples. A large number of measurements were made in shallow water areas with water depths ranging from 50 to 160 m. The statistical distributions of the derived bottom-loss values are found to be nearly invariant to various sediment properties derived from the corresponding core samples. These properties include seafloor sound speed, density, and porosity as well as a number of others. It is shown from Parabolic Equation model calculations that use Hamilton's values for the different sediment types [J. Acoust. Soc. Am. **68**, 1313 (1980)] that the TL can vary by 15 dB or more at a range of 20 km. Therefore the absence of substantial bias in the data with respect to sediment types is somewhat surprising and worth investigation. The possible explanations, such as high spatial variability of seafloor processes including presence of ripples, gas, and other causes are being investigated and will be discussed. [Approved for public release].

8:40

5aUW3. High-frequency sediment sound speed and attenuation measurements during TREX13 (Target and Reverberation Experiment 2013) with a new portable velocimeter. Laurent Guillon (Ecole navale/IRENav, BCRM Brest, CC600, Brest cedex 9 29240, France, laurent.guillon@ecole-navale.fr), Xavier Demoulin (Maree, Ploemeur, France), Brian T. Hefner (Acoust.Dept., Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Dapeng Zou (School of ElectroMech. Eng., Guangdong Univ. of Technol., Guangzhou, China)

During the Target and Reverberation Experiment 2013 (TREX13), high-frequency measurements of sediment sound speed and attenuation were collected throughout the experiment site. These measurements were performed using the INSEA, a diver-portable array of sources and receivers conceived and developed by French companies in collaboration with research institutions. During each deployment of the instrument, the INSEA was inserted 10–15 cm into the sediment and narrow-band pulses covering the 70 to 350 kHz range were transmitted through the sediment. The sound speed is determined from the time-of-flight and attenuation is determined from the amplitude ratio of the transmissions through the sediment and through the water. The variability of the TREX13 site made it possible to collect data in several different sediment types including mud, silty-sand, and sand sediments each with low to high concentrations of shells. In addition to the acoustic measurements, diver cores and shell samples were also collected. The sound speed and attenuation measured in these sediments are presented and discussed. [Work supported by DGA, ONRG, and SERDP.]

Contributed Papers

9:00

5aUW4. Journey to Antarctica: Modeling crustal structure with an earthquake and a genetic algorithm. Priscilla Brownlow (Graduate Program in Acoust., Penn State Univ., 307B Dunham Hall, White Course Apt., University Park, PA 16802, pdb153@psu.edu), Richard Brazier, and Andrew Nyblade (Dept. of GeoSci., Penn State Univ., University Park, PA)

In a previous work, we have used the genetic algorithm NSGA-II to generate a set of solutions to model the receiver functions and dispersion curves of several seismometer stations located in southern Africa. Now in continuation of applying the NSGA-II to seismic problems, we have used it to model the average velocity profiles along two-dimensional paths from a single seismic event to several stations across West Antarctica. The event was a rare continental earthquake of magnitude 5.6 that took place in West Antarctica near the Ross Ice Shelf during the austral winter of 2012. Data were collected from stations in the Global Seismic Network as well as a local network during the 2012–2013 field season. The seismograms were first modeled using a full body wave modeling code that generates synthetics based on a structure composed of layers with user-defined velocities, thicknesses, and densities. Those models then served as the starting models in NSGA-II, which created a set of solutions from which an average structure with error bounds was calculated for each station.

9:15

5aUW5. Physics-based inversion of multibeam sonar data for seafloor characterization. Brian T. Hefner, Darrell R. Jackson, Anatoliy N. Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu), and Gorm Wendelboe (Teledyne-RESON A/S, Slangerup, Denmark)

As part of a continuing effort to develop a physics-based seafloor inversion technique, both acoustic and environmental data were collected during the Target and Reverberation Experiment 2013 (TREX13). The data were collected along a 350 m long survey track that sampled several sediment types including sand, silty-sand, and mud. A RESON 7125 multibeam sonar was modified to collect data along the track from 150-450 kHz in 50 kHz intervals. Ground-truth data on seafloor properties were acquired along this track, including measurements of roughness, sound speed, attenuation, and both discrete and continuous volume heterogeneity. A model was used to generate echo intensity time series including scattering by both seafloor roughness and volume heterogeneity. Model-data fits were used to provide estimates of acoustic attenuation, volume scattering strength, and roughness spectral parameters. Volume scattering is treated using an empirical model, while roughness scattering is treated using the small-slope approximation. [Work supported by SERDP.]

9:30

5aUW6. Low frequency sound attenuation measurements in marine sediments. Ross Chapman (Univ. of Victoria, 3800 Finnerty Rd., VICTORIA, BC V8P5C2, Canada, chapman@uvic.ca)

This paper reports measurements of sound attenuation in marine sediments from two locations on the outer New Jersey continental shelf. At one site the sediment in the top 20 m is primarily sandy clay, while the other site includes a thin (3–5 m), over-lying layer of sand at the sea floor. The attenuation was inverted from close range, broadband data from light bulb implosions deployed at stations at the sites. The inversion method made use of the time-frequency dispersion information in signals received at single hydrophones. The signals were first processed by time warping to resolve the propagating modes at relatively close ranges (50–80 water depths). The inversion is carried out in two stages. The first stage inverted the sound speed and density by modeling the modal group velocities, and these estimates were used in the second stage to invert the attenuation from the modal amplitude ratios. The results provide estimates of low-frequency sound attenuation that can be compared to predictions from different models of sound propagation to assess the frequency dependence in the band from 100–500 Hz.

9:45

5aUW7. Seafloor sound-speed profile and interface dip angle measurement by the image source method. Samuel Pinson and Charles W. Holland (Penn State Univ., Appl. Sci. Bldg., Rm. 202a, State College, PA 16802, samuelpinson@yahoo.fr)

The image source method is an efficient way to perform a sound-speed tomography for seafloor characterization. To date, however, it has been limited by a locally range-independent approximation for layer boundary geometry. In other words the layer boundary had to be parallel and flat within 1 Fresnel zone of the measurement system. Here, the method is extended to take into account realistic variations of interface dip angles. To do so, the elliptical wavefront shape approximation of the reflected waves is used. This permits a fairly simple equation relating travel time to the sine of the dip angle, and consequently to an equation for the equivalent medium sound-speed. The Radon transform is exploited to extract this dip angle parameter. Simulations with varying layer dip angles and curvature provide insight into the strengths and limitations of the method.

10:00–10:15 Break

10:15

5aUW8. Sensitivity analysis of the image source method to roughness and volume heterogeneities. Samuel Pinson and Charles W. Holland (Penn State Univ., Appl. Sci. Bldg., Rm. 202a, State College, PA 16802, samuelpinson@yahoo.fr)

In the context of the sediment characterization, the image source method provides a fast and automated sound-speed profile measurement of the seafloor. This technique is based on the analysis of the seafloor reflected acoustic wave as a collection of image sources whose positions are linked with the thicknesses and the sound speed of the sediment stack. The presence of interface roughness and volume inhomogeneities will reduce phase coherence between the receivers and thus may reduce the ability to precisely obtain the image source position. However, “blurring” of the image source position may provide useful clues about the roughness and/or volume heterogeneities. Recent measurements were obtained using an Autonomous Underwater Vehicle (AUV) towing a broadband source (frequency band from 1600 to 3500 Hz) and a linear array of hydrophones. Based on that configuration, a sensitivity study of the effect of roughness and volume heterogeneities on the image source method is presented and discussed.

10:30

5aUW9. Reflection-coefficient inversion for compressional- and shear-wave dispersion in porous and elastic layered media. 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., The Penn State Univ., State College, PA), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

This paper considers Bayesian inversion of seabed reflection-coefficient data for the compressional- and shear-wave velocity dispersion and

attenuation-frequency dependence in arbitrarily layered porous and elastic media. The seabed is modeled using Buckingham's viscous grain shearing model which obeys causality. Seabed layers are parametrized in terms of six fundamental parameters, including thickness, porosity, compressional and shear grain-to-grain moduli, material exponent, and the compressional visco-elastic time constant. These fundamental parameters are used to compute density, compressional- and shear-wave dispersion curves, and compressional and shear attenuation-frequency curves for each layer. The curves are used in the inversion to predict spherical-wave reflection coefficients as a function of frequency (300–3000 Hz) and grazing angle (12–75 degrees), which include the effects of shear waves in arbitrarily layered media. In addition, the seabed layering is estimated from the data by applying a trans-dimensional Bayesian model. The ability to resolve shear-wave velocity and attenuation structure is studied using simulated data. Finally, compressional- and shear-wave dispersion are presented and discussed from measured reflection data at a sandy site in the Tyrrhenian Sea. [Work supported by ONR Ocean Acoustics.]

10:45

5aUW10. Inversion of shear wave speed in coastal sediments using interface waves. Gopu R. Potty, Jennifer Giard, James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), Benjamin Goldsberry, and Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Shear speeds in semi-consolidated and consolidated shallow water sediments can significantly impact compressional wave attenuation and arrival times of acoustic normal modes. In addition shear properties of sediments are directly related to the strength of the sediments in geotechnical applications. All of these factors emphasize the importance of estimating shear speeds in shallow water sediments. One of the most promising approaches to estimate shear speed is to invert the shear speed profile using the dispersion of interface waves (Scholte waves). Interface wave data from a small scale experiment conducted in very shallow water in coastal Rhode Island will be presented. The University of Rhode Island's shear measurement system consisting of vertical axis and 3-axis geophones were used to collect data in 3 m of water. Interface waves were excited by dropping a weight from a research vessel. Modeling of interface waves will be carried out using Finite Element Method (FEM) and a dynamic stiffness matrix model. Sediment properties will be inferred based on the modeling and data-model comparison. The estimated sediment properties will be compared with historic core data from the field test location. [Work supported by Office of Naval Research.]

11:00

5aUW11. Laboratory measurements of shear wave properties in marine sediments using bender element transducers. Kevin M. Lee, Megan S. Ballard, Sanjai Bashyam, and Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu)

In shallow water, acoustic propagation is often controlled by the properties of the seabed. In environments where the shear speed in the sediment approaches the sound speed in the water column, wave conversion at the bottom has been identified as a dominant loss mechanism. The ultimate goal of this work is to develop a device for measuring both compressional and shear wave properties *in* situ in marine sediments. This work presents laboratory measurements using bender element transducers to measure shear wave properties in marine sediments. The transducer consists of two long, thin piezoceramic plates rigidly bonded along their lengths driven 180 degrees out of phase in the length extensional mode so that the differential change in length of each plate causes the composite element to bend [J. Acoust. Soc. Am **63**, 1643–1645 (1978)]. When the transducer is embedded in a medium, mechanical motion is transferred from the bender to the particles in the medium in a manner such that particle motion is perpendicular to the length dimension of the element. Laboratory measurements demonstrate bender sensitivity to low amplitude shear waves in sandy and muddy sediments. [Work supported by ARL:UT IR&D.]

11:15

5aUW12. An inverse method for estimating sediment sound speed. Tao Lin and Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ 07102, tl48@njit.edu)

A fast approach for solving the inverse problem of estimating sediment sound-speed based on the Deift-Trubowitz trace formula is being investigated in our research. Under certain assumptions, this algorithm can recover the sound speed profile in the seabed using pressure field measurements in the water column at low frequencies. The inversion algorithm, employing a modified Born approximation, works well with synthetic data. Results are compared to those of previously developed methods and demonstrate improvement especially at sharp changes in sound speed. Although the method is stable and effective with noise-free data, problems arise when noise is considered. In our work, we develop regularization methods to remedy this problem. Finally, we recognize that some assumptions necessary for this algorithm to work may not be realistic; we discuss ways to relax these limitations. [Work supported by ONR.]

5a FRI. AM

Session 5pMU

Musical Acoustics and Structural Acoustics and Vibration: Computational Methods in Musical Acoustics II

Edgar J. Berdahl, Chair
Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803

Invited Papers

1:00

5pMU1. Real-time physical models of musical instruments: Applications and findings. Florian Pfeifle and Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstrasse 13, Hamburg 20354, Germany, Florian.Pfeifle@uni-hamburg.de)

Real-time auralization of physical models has attracted vivid interest over the last years. This is mainly due to the rising computational capabilities of personal computers and the accessibility of specialized (external) accelerating hardware, like GPGPUs (general-purpose graphics processing units) or FPGAs (field programmable gate arrays). In this work, an extended framework of real-time physical models of musical instruments, calculated with symplectic and multi-symplectic finite difference algorithms, on a FPGA is presented. The former study, as presented in earlier publications by the authors, is extended in three aspects: (a) A methodology for coupling FPGA boards via a highspeed general purpose IO, to facilitate calculations of larger instrument geometries, such as piano sound-boards, is implemented. (b) A generalized design structure for all models is developed. (c) An enhanced external interface communication protocol is realized. These extensions resulted in several new possible applications for music and for musicological research.

1:20

5pMU2. Embedded physical modeling synthesis in three dimensional environments. Stefan Bilbao (Music, Univ. of Edinburgh, Rm. 7306B, JCMB, Kings Bldgs, Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, sbilbao@staffmail.ed.ac.uk)

3D audio rendering of virtual spaces, for purposes of artificial reverberation, or in concert hall auditioning has seen great advances in recent years. Of particular interest are wave based techniques, such as finite difference time domain methods. Such methods are computationally intensive, and parallel architectures, such as GPGPUs, can be of use in accelerating computation times. A further use of such methods is in synthesis—through the embedding of physical models in a three dimensional space, allowing the complete spatial rendering of the acoustic field. In this paper, a variety of membrane- and plate-based percussion instruments will be discussed, with special emphasis on implementation issues in parallel hardware. Sound examples will be presented.

1:40

5pMU3. Computation and simulation of frequency variations in musical instrument sounds. James W. Beauchamp (School of Music and Dept. of Elec. & Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr, Urbana, IL 61801-6824, jwbeauch@illinois.edu)

Frequency variations of musical instrument sounds were measured using phase-derivative and frequency-tracking methods based on the short-time Fourier transform. Frequency variations are important features of instrument sounds and are very useful for musical expression. Three categories of variation are: vibrato, portamento, and microvariation. Microvariations exist even when a tone is played at a constant pitch, and they can be approximated as small frequency-deviation low-frequency noise signals. Portamento is a purposeful pitch glide embellishment that can occur during attacks, between notes, or, less often, at the ends of notes. Vibrato can be characterized as an approximately sinusoidal frequency variation, and usually its amplitude is sufficient to interact with instrument resonances and cause significant harmonic amplitude modulations. Deviation amplitudes and frequencies of acoustic instrument vibratos are not perfectly steady, but rather vary over the durations of instrument tones. Measurements of vibrato characteristics of the harmonic frequencies and amplitudes as well as the frequency and amplitude microvariations of various instruments and voice indicate that a variety of parameters are required for effective instrument synthesis. The challenge in synthesis is to avoid a "mechanical sound."

Contributed Papers

2:00

5pMU4. Haptic interaction design using Synth-A-Modeler. Edgar J. Berdahl (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

Synth-A-Modeler is an open-source and modular software environment for designing physical models using a mechanical analog approach. Notably, physical models provide the most reliable method for programming haptic force-feedback interactions that can be ported across a wide array of haptic devices and usage scenarios. In this presentation, we explain how Synth-A-Modeler facilitates teaching haptic interaction design, with an emphasis on audio-haptic interaction. A series of example models demonstrates how mass-interaction, modal synthesis, and digital waveguide elements, as well as combinations thereof, can be employed in

Synth-A-Modeler to simulate virtual audio-haptic environments. Although Synth-A-Modeler can hide the details of the model implementations, some equations are employed to calibrate the models. The models are tested with the FireFader open-source haptic device; however, the models should be compatible with a wide array of other haptic devices and DSP targets as well.

2:15

5pMU5. Physical modeling of musical instruments on handheld mobile devices. Gregory P. Scandalis, Julius O. Smith, and Nick Porcaro (moForte.com, 286 Carmelita Dr., Mountain View, CA 94040, gps@moforte.com)

Handheld mobile computing devices are now ubiquitous. These devices are powerful, connected and equipped with a variety of sensors. Their pervasiveness has created an opportunity to realize parametrically controlled, physically modeled, virtual musical instruments. We will present a brief history of physically modeled musical instruments and the platforms that those models have been run on. We will also give an overview of what is currently possible on handheld mobile devices including modeling done in the "moForte Guitar" mobile application. "moForte Guitar" is an application for mobile devices that models the physics of the guitar family of instruments. Modeling makes expressive interactive articulation possible, which cannot be directly achieved with sample playback techniques. Features that are modeled include: electric and acoustic instruments, strumming at various string positions, string scraping and damping, harmonics, glissando, automated modeling of strumming, statistical variation of physical parameters, feedback/distortion and classic processed electric guitar effects. We will show a number of real-time demonstrations on a handheld mobile device for what is possible with this model.

2:30

5pMU6. Spectrally accurate numerical solution of acoustic wave equations. John W. Amuedo (Signal Inference, 3267 Butler Ave., Los Angeles, CA 90066, jamu@siginf.com)

Finite difference models of wave propagation have presented challenging problems of stability and accuracy since initial experimentation with these models began on early digital computers. The advent of spectral methods in the late 1960s has led to the latter's increasing use for solving differential equations in a range of fluid dynamic, electromagnetic and thermal applications. Spectral methods transform a physical grid of state variables (such as acoustic velocity and pressure) into an alternative spectral space characterized by a particular set of basis functions. Spatial derivatives of physical state variables are computed in spectral space using exact differential operators expressed in terms of those functions. Fast numerical transforms are employed to exchange immediate state of a simulation between its spectral and physical representations. In problems equally suited to spectral and finite difference formulation, spectral methods often yield increased fidelity of physical results and improved stability. Spectral methods sometimes enable computational grid size requirements of a simulation to be substantially reduced, with concomitant computational savings. This paper reports on spectral implementations of the acoustic wave equation and Webster horn equation for simulating audio transducer cavities, musical instrument resonators, and the human vocal tract.

2:45

5pMU7. The mother tongue of organ pipes-Synchronization, experiments, numerical simulations, and model. Jost Fischer and Markus Abel (Dept. for Phys. and Astronomy, Univ. of Potsdam, Karl-Liebknecht-Str 24/25, Potsdam, Brandenburg 14476, Germany, jost.fischer@uni-potsdam.de)

We present recent results on the synchronization (Mitnahme Effect) of organ pipes. Previous work has focused on the detailed measurement and reconstruction of the driving of an organ pipe by a loudspeaker. As a result the full Arnold tongue was measured and reconstructed and a synchronization could be found down to a fraction of 1/500 of the sound pressure level of the organ pipe. In this contribution, we give detailed results on the experimental determination of the Arnold Tongue for two pipes. The results are accompanied by detailed numerical simulations of sound generation and sound radiation with the aim to clarify the interaction of the jets and the dependence of the synchronization region on coupling strength (realized by varying distance). Furthermore, we propose a model for the coupling function.

3:00

5pMU8. Towards a physical model of the berimbau: Obtaining the modal synthesis of the cabaza. Pablo Castellanos Macin and Julius O. Smith (Dept. of Music - Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, pablocm@ccrma.stanford.edu)

The worldwide presence of Brazilian culture grows every day. However, some of the musical instruments used in its principal cultural activities lack of a formal acoustic analysis which would make them more understandable for the rest of the world. One of them is the berimbau-de-barriga (berimbau), which consists of a string (wire) attached to an arched rod and a resonance box called cabaza. Modeling the berimbau will not only open up possibilities for its application to other musical genres, but will also allow the incorporation of its characteristics into new virtual instruments. The present work describes the modal synthesis of the cabaza, i.e., modeling this sounding box as a parallel bank of digital resonators. Impulse response measurements were obtained using a force hammer, and second-order resonator frequency-responses were fit to the data using MATLAB.

3:15

5pMU9. Aural ordinary differential equations: Methods for generating audio from mass-spring systems. Andrew S. Allen (UCSD, 4757 Clairemont Mesa Blvd., Apt. 306, San Diego, CA 92117, drewbitllama@gmail.com)

In this article, I focus on the harmonic oscillator as a model by which to compare several numerical methods for solving ordinary differential equations (ODEs). I first define the simple harmonic oscillator as an ODE and then extend its behavior by adding additional forces and physical properties to the equation. I next proceed to discuss computational methods for solving ODEs and use the oscillator model as a means of evaluating and comparing three methods in terms of their stability, drift, and computational costs when working at the audio-rate.

3:30

5pMU10. Modeling the free vibrations of an acoustic guitar top plate. Micah R. Shepherd, Stephen A. Hambric, and Dennis B. Wess (Appl. Res. Lab., Penn State Univ., PO Box 30, M.S. 3220B, State College, PA 16801, mrs30@psu.edu)

Using computer models to simulate the sound of an acoustic guitar can significantly decrease lead time for new designs. In order to create an accurate computer model of a Dreadnought-style acoustic guitar, a sequential modeling approach was used. A finite element model of a bare top plate with braces and a bridge plate was created. The top and plate and braces were modeled as plate elements with orthotropic material properties. The natural variation of the wood properties was also examined along with their dependence on moisture content. The modes of the model were then compared to experimentally obtained modes from top plate prototypes. The modeshapes of the model compared well to those measured. Uncertainty analysis was also performed and the statistical bound of natural error between wood samples was determined to be approximately 8%. The natural frequencies of the model fell within the error bound for lower-order modes but diverged slightly for several higher-order modes. These results indicate the importance of using accurate material properties in models of acoustic guitars.

Session 5pSC

Speech Communication: Crosslinguistic Analysis (Poster Session)

Molly E. Babel, Chair
Linguist., Univ. of British Columbia, 2613 West Mall, Totem Field Studios, Vancouver, BC V6T 1Z4, Canada

Contributed Papers

5pSC1. Range of variability in native and non-native spontaneous speech intervocalic stops. Miguel Simonet (Spanish and Portuguese, Univ. of Arizona, Tucson, AZ), Natasha L. Warner, Dan Brenner, Maureen Hoffmann, Andrea Morales, and Alejandra Baltazar Molina (Dept. of Linguist., Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028, nwarner@u.arizona.edu)

Speakers produce sounds differently in spontaneous vs careful speech, and how they do this shows both similarities and differences across languages. The current project examines spontaneous conversational speech and read speech among monolingual English speakers, Dutch-English bilinguals, and Spanish-English bilinguals (for Dutch and Spanish, in both their L1 and English). The phonology of intervocalic stops differs in these languages: Dutch has final devoicing, Spanish has approximation of /bdg/, and English has flapping of /td/. In our recordings, Dutch speakers often devoiced final stops in both Dutch and English in spontaneous speech, while native English speakers produced voiced stops or approximants. Speakers of all the languages produced some approximant realizations and some deletions. Through measurements of consonant duration, amplitude dip during the consonant, and cessation of voicing, this work shows the range of acoustic variability produced by speakers of three languages in their L1 and their L2, in spontaneous and careful speech. This allows a comparison of how much of speech variability stems from the native language phonology, from language-specific phonetics, and from language-general spontaneous speech reduction.

5pSC2. Intelligibility of speaking styles elicited by various instructions. Rachael C. Gilbert, Nicholas Victor (Linguist, Univ. of Texas at Austin, 4812 Ave. H, Apt. B, Austin, TX 78751, rachaelgilbert@gmail.com), Bharath Chandrasekaran (Commun. Sci. & Disord., The Univ. of Texas at Austin, Austin, TX), and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

The acoustic-phonetic modifications made by talkers are attuned to the specific communicative situations that listeners are experiencing (Lam and Tjaden, 2013; Hazan and Baker, 2011). The extent to which such modifications are under explicit control remains largely unknown. This study examined the extent to which native and non-native talkers can implement acoustic-articulatory enhancements following specific instructions and the extent to which these changes will improve intelligibility. Ten native and 10 Korean-accented talkers read sentences in various styles including in conversational and clear speech, while imitating a native speaker's conversational and clear speech, with exaggerated vowels, more slowly, and more loudly. Sentences were mixed with noise (−5 dB SNR) and presented to native listeners. Intelligibility results revealed that nonnative talkers were overall less successful in enhancing intelligibility following different instructions compared to native talkers. Instructions to speak clearly and imitating native clear speech sentences provided the largest intelligibility benefit while the instructions to slow down were least successful in improving intelligibility across talkers. Speaking loudly and exaggerating vowels increased intelligibility only for native talkers. Acoustic analyses will examine which acoustic-phonetic changes were implemented following each instruction. The results have important implications for enhancing intelligibility in difficult communicative situations (e.g., classrooms).

5pSC3. Normalization and matching routine for comparing first and second language tongue trajectories. Shusuke Moriya, Yuichi Yaguchi, Naoki Terunuma, Takahiro Sato, and Ian Wilson (Univ. of Aizu, Tsuruga Ikkimachi, Aizuwakamatsu, Hukushima 965-8580, Japan, s1190242@gmail.com)

The main purpose of this research is specifying the articulation difference between L1 and L2 speakers by digitizing tongue motions and analyzing their differences between utterances. Differences in tongue motion directly influence speakers' pronunciation, so it may be possible to determine a speaker's L1 from tongue motion data. By comparing L1 and L2 speakers' tongue motion, we can also guide L2 speakers to improve their L2 pronunciation. In this research, we use coronal cross sections of the tongue taken by an ultrasound scanner to carry out the following: first, record the ultrasound of a speaker's tongue motion using the story "The Boy Who Cried Wolf." Then, sample mobility information by using histogram of oriented gradients. Next, use Karhunen-Loeve expansion to reduce the vector dimensions. At this time, we get the average difference between the starting vector of tongue motion and the subsequent vectors, then normalize the direction of the two averages. Finally, we use dynamic time warping to compare each vector per frame. The experiment results allowed us to compare speakers' tongue mobility information in words which were recorded in different experiment environments or by different speakers.

5pSC4. Coarticulatory effects of lateral tongue bracing in first and second language English speakers. Sunao Kanada, Ian Wilson (CLR Phonet. Lab., Univ. of Aizu, Tsuruga, Ikki machi, Aizuwkamatsu, Fukushima 965-8580, Japan, s1180011@u-aizu.ac.jp), Bryan Gick (Dept. Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada), and Donna Erickson (Showa Music Univ., Kawasaki, Japan)

This study uses electromagnetic articulometry (EMA) to examine the coarticulatory effects of tongue bracing in L1 and L2 English speakers. The tongue is hydrostatic, so we brace it against our teeth for added control, and this bracing is an important part of pronunciation. The amount of bracing may differ across languages (and may be part of one's articulatory setting), so understanding these differences could be a key to L2 pronunciation learning. Although lingual coarticulation has been examined using acoustics and midsagittal views of the vocal tract, not much focus has been placed on the coronal view. We collected EMA point-tracking data from two native speakers of North American English and looked at the movement of a lateral tongue marker. As stimuli, we choose the nursery rhyme "Mary had a Little Lamb" because of the variation in vowels, and also the /l/ and /r/ phonemes, which are absent in Japanese. Initial results show differences between vowels that occur next to /l/ and those that occur next to /r/ and stops. Results will also be presented for Japanese speakers of both their L1 (Japanese) and L2 English. If we find crosslinguistic differences in bracing, this fact will be important for pedagogical purposes.

5pSC5. Perceptual and phonotactic effects in loanword adaptation: English postvocalic stops in Taiwan Mandarin. Jui Ching Chang (National Chiao Tung Univ., 3/F, Humanities Bldg. 2, 1001 Ta-Hsueh Rd., Hsinchu 300, Taiwan, Hsinchu 300, Taiwan, showtheray@gmail.com)

When an English word with a postvocalic stop is borrowed into Taiwan Mandarin (TM), because TM allows only nasal coda consonants, an often-used strategy to repair the illegality is vowel-insertion (e.g., Wood → [wu.tɤ]). Based on a corpus study of 335 English borrowed names, this trend is confirmed (76%). Among the deleted cases, an asymmetry of different places of articulation was found: Coronal stops are deleted most often (15%, e.g., Hollywood → [hau.lai.wu]), and dorsals more often than labials (12%, e.g., Titanic → [tʰiɛ.ta.ni] vs 0%, e.g., Jeep → [tɕ. pʰ**u**]). Following Kang's (2003) perceptual explanation, the tendency of coronal-deletion can be explained by the fact that postvocalic coronals are often unreleased and thus less perceptually salient to TM speakers. According to TIMIT corpus, the release rate of coronals, labials, and dorsals stops is 37%, 52%, and 83%, respectively (Kang, 2003). However, this cannot explain the reversed pattern of dorsals and labials. I propose that this is due to another factor: Labial coda is marked in TM since only [n] and [ŋ], but not [m], can occur in coda position. In other words, the deletion of postvocalic stops depends on the saliency that considers both perceptual and phonotactic factors.

5pSC6. Neural plasticity in phonetic training of the /i-I/ contrast for adult Chinese speakers. Bing Cheng (Xi'an Jiaotong Univ., 164 Pillsbury Dr. SE, 115 Shevlin Hall, Minneapolis, Minnesota 55455, chengbing72@gmail.com) and Yang Zhang (Univ. of Minnesota, Minneapolis, MN)

This study investigated neural plasticity associated with phonetic training using a software program developed after Zhang *et al.* [NeuroImage **46**, 226–240 (2009)]. The target sounds were /i/ and /I/ in English, a non-phonemic contrast in Mandarin Chinese. The training program integrated four levels of spectro-temporal exaggerations, multi-talker variability, audio-visual presentation, and adaptive listening in seven sessions, each lasting about 15 min. The participants were ten adult Chinese English-as-a-second-language learners. Identical pre- and post-tests were administered one week before and after training. Behavioral measures included discrimination and identification tasks as well as formant analysis of vowel production. Event Related Potential (ERP) measures examined training-induced changes in the mismatch negativity (MMN) responses. The behavioral results showed significant improvement in identification and discrimination scores and a clear continuous-to-categorical perceptual shift, which were also reflected in the MMN responses for detecting the across- vs within-category differences at the pre-attentive level. There was also strong evidence for transfer of learning from trained to untrained stimuli as well as from perception to production. The results demonstrate the existence of substantial neural plasticity for speech learning in adulthood and provide further testimony for the efficacy of the adaptive audiovisual training method for promoting second language phonetic learning.

5pSC7. The perception of Mandarin lexical tones by native Japanese and Thai listeners. Kimiko Tsukada (Int. Studies, Macquarie Univ., Sydney, NSW, Australia), Rungpat Roengpitya (Faculty of Liberal Arts, Mahidol Univ., Mahidol University, Bangkok, Thailand, rungpat@gmail.com), Hui Ling Xu (Int. Studies, Macquarie Univ., Sydney, NSW, Australia), and Nan Xu (Linguist, Macquarie Univ., Sydney, NSW, Australia)

Mandarin differentiates four tones (T1: high level (ā), T2: high rising (á), T3: dipping (ǎ), T4: high falling (à)). Learning these lexical tones is known to be difficult for those from non-tonal language background (e.g., English). What about listeners with no knowledge of Mandarin but have varying experience with tones or pitch variation? This study examined the discrimination of 6 Mandarin tone contrasts (T1-T2, T1-T3, T1-T4, T2-T3, T2-T4, and T3-T4) by native speakers of Japanese (pitch-accent language) and Thai (tonal language). The listeners' tone discrimination accuracy was assessed in a categorical discrimination test developed by Jim Flege and colleagues. Both non-native groups were less accurate than the native group, in particular, for the T1-T2, T1-T3, T1-T4, and T2-T3 contrasts. Despite using lexical tones in their first language (L1), Thai listeners did not have a distinct advantage over Japanese listeners and the two groups showed a similar pattern of results. Overall, discrimination accuracy of contrasts involving T1 was lower than other contrasts with the exception of T2-T3. Both Japanese and Thai listeners had greatest difficulty with T2-T3. Since previous knowledge of L1 tones may interfere with the perception of non-native tones, these results will be discussed with reference to a Thai tonal system.

5pSC8. The effect of language distance and language experience in third language acquisition. Seung-Eun Chang (East Asian Lang. and Cultures, Univ. of California Berkeley, 3413 Dwinelle, Berkeley, CA 94720-2230, sechang71@berkeley.edu)

This research aims to examine the role of second-language (L2) phonology in third-language (L3) acquisition. As a mean to assess the degree of influence of the L1 accent and L2 accent in L3 production, an experiment that involved the judgment of a foreign accent was developed. Two groups of native English speakers [(i) five who had not learned any language other than Korean, and (ii) five who had learned Japanese before learning Korean] produced Korean sentences, and 25 native Korean-speaking raters identified each production according to the speaker's dominant accent, either English or Japanese. The results revealed that native English speakers who had learned Japanese before learning Korean were more frequently identified as having a strong Japanese, rather than English, accent in their Korean production. In accounting for the results, several hypotheses were discussed, including language distance (typological proximity), inherently different mechanisms for foreign language acquisition as compared with the natural acquisition of the L1, psycho-affective factors, and stronger links between foreign languages in the speaker's mind. The findings of this study provide further evidence for the claim that L2 exerts an influence on L3 accent; however, this interference is reduced with an increase in L3 proficiency

5pSC9. The effects of acoustically enhanced speech on lexical tone perception in Mandarin as second language learners. Huei-Mei Liu (Special Education, National Taiwan Normal Univ., 162 Ho-Ping East Rd. SEC 1, Taipei 106, Taiwan, liumei@ntnu.edu.tw) and Feng-Ming Tsao (Psych., National Taiwan Univ., Taipei, Taiwan)

The tonal language learners who speak non-tone language have difficulty discriminating lexical tones of a tone language. This study aimed to examine the effects of acoustically enhanced speech on perceptual sensitivities and organizations of lexical tones in Mandarin as second language learners. Three groups of participants were recruited, native Mandarin speakers ($n = 26$), native English speakers ($n = 28$), and native Thai speakers ($n = 26$). Both groups of Mandarin learners have learnt Mandarin as second language (L2) for several years. Mandarin lexical tone discrimination and identification tasks with two sets of tone stimuli, with and without pitch contour exaggeration, were used in this study. The results showed that Mandarin L2 learners performed less well on the tone discrimination and identification tasks, relative to native Mandarin speakers. In addition, Mandarin L2 learners perceptually weight less to pitch direction than pitch height in their perceptual organization for tones, showing different perceptual weights from native Mandarin speakers. In the context of listening to acoustically enhanced stimuli, the group difference on tonal sensitivity and cue-weighting patterns of perceptual organization were greatly reduced. Results imply that the signal enhancement facilitates Mandarin L2 learners to process lexical tones.

5pSC10. Acoustic measurement of word-initial stop consonants in English-French interlingual homophones. Paula Castonguay and Jean E. Andruski (Commun. Sci. and Disord., Wayne State Univ., 60 Farnsworth St., 207 Rackham Bldg, Detroit, MI 48202, dx4720@wayne.edu)

The purpose of the present study is to examine word-initial stop consonants of Canadian English (CE) and Canadian French (CF) interlingual homophones in order to describe how they differ in their acoustic properties. Interlingual homophones (IH) are words across languages that are phonemically identical but phonetically and semantically different, for example, English two /tu/ and French tout <all> /tu/. Even though they are deemed phonemically identical, at the acoustical level they may be quite different. In the current study, Canadian bilingual English and French speakers were asked to produce interlingual homophones embedded in carrier phrases and in isolation. Voice onset time, relative burst intensity, and burst spectral properties of the IH words were measured and compared within and across languages. The acoustic measurements obtained will be used (1) to make predictions about which acoustic features may provide cues to language identity, and (2) to create stop tokens for a Goodness Rating study. Results from this study will provide insight on the acoustic-phonetic representation of stop consonants in Canadian bilingual English and French speakers.

5pSC11. The use of durational variables to characterize the rhythmic patterns of non-fluent Japanese utterance by non-native speakers. Shigeaki Amano, Kimiko Yamakawa (Faculty of Human Informatics, Aichi Shukutoku Univ., 9 Katahira, Nagakute, Aichi 480-1197, Japan, psy@asu.aasa.ac.jp), and Mariko Kondo (School of Int. Liberal Studies, Waseda Univ., Shinjuku, Tokyo, Japan)

Twenty-nine durational variables were examined to clarify rhythmic characteristics in non-fluent Japanese utterances by non-native speakers. Discriminant analysis with these variables was performed on 343 Japanese words, each pronounced in a carrier sentence by six native Japanese speakers and 14 non-native Japanese speakers (7 Vietnamese with low Japanese proficiency and 7 Chinese with high Japanese proficiency). The results showed that a combination of two durational variables could discriminate Japanese speakers from Vietnamese speakers with a small error (8.7%, n = 4458), namely the percentage of vowel duration and the average of "Normalized Voice Onset Asynchrony," which is an interval time between the onset of two successive vowels divided by the first vowel's duration. However, these two variables made a large error (39.4%, n = 4458) in the discrimination of Japanese speakers from Chinese speakers who had higher Japanese proficiency than Vietnamese speakers. These results suggest that the two variables characterize the rhythmic pattern in a non-fluent Japanese utterance by non-native speakers with low Japanese proficiency. [This work was supported by JSPS KAKENHI Grant Numbers 22320081, 24652087, 25284080, and by Aichi Shukutoku University Cooperative Research Grant 2013-2014.]

5pSC12. Prosodic characteristics in Japanese speech by Taiwan Mandarin speakers and native Japanese speakers. Naomi Ogasawara (Ctr. for Lang. Res., Univ. of Aizu, 90 Tsuruga Ikkimachi, Aizuwakamatsu 965-8580, Japan, naomi-o@u-aizu.ac.jp), Timothy J. Vance (National Inst. for Japanese Lang. and Linguist, Tokyo, Japan), and Chia-Lin Shih (The Graduate Inst. of Linguist, National Taiwan Normal Univ., Taipei, Taiwan)

Previous studies (Ishihara *et al*., 2011; Sato 1995) show that prosody contributes more to native-like accents than segments do. It was also found that compared with errors in timing, errors in pitch accent in Japanese speech were more tolerable to native and non-native speakers. This suggests that non-native speakers pay less attention to pitch accent when speaking Japanese; as a result, their acquisition of correct pitch accent does not progress as their overall Japanese proficiency improves. In this study, Taiwan Mandarin speakers and native Japanese speakers produced Japanese words with different syllable structures, some containing all short syllables and others at least one long syllable. These words are 2 to 4 moras long and have nine pitch accent patterns. Each participant produced each word in isolation and in a carrier sentence. All speech data were acoustically analyzed to measure (1) the highest F0 point in accented syllables and (2) the difference in F0 between an accented syllable and adjacent unaccented syllables. The purpose of this study is to investigate common F0 patterns in pitch accents among native and non-native speakers of Japanese, and common pitch accent errors made by the non-native speakers.

5pSC13. Vowel identification in temporal modulated noise for native and non-native listeners: Effect of language experience. Jingjing Guan, Chang Liu (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, jane.guan@utexas.edu), Sha Tao, Lin Mi, Wenjing Wang, and Qi Dong (State Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Beijing, China)

Previous work in our laboratories found vowel identification in babble was significant different between Chinese-native listeners in China and United States. As a follow-up, the current study focused on whether the two groups of Chinese listeners had any difference in using temporal cues of noise for vowel identification. Temporal modulation transfer function and vowel identification in temporal modulated noise were measured for American English native (EN) listeners, Chinese-native listeners in United States (CNU), and Chinese-native listeners in China(CNC). Results revealed that TMTF is similar across three groups, indicating that psychophysical temporal processing is independent of listeners' language backgrounds. However, for vowel identification in noise, EN and CNU listeners showed significantly greater masking release from temporal modulation of the noise than CNC listeners, especially at low SNR conditions (e.g., −12 dB). Altogether, native English exposure may change the use of temporal cues on English vowel identification for Chinese-native listeners.

5pSC14. Influence of second language on the perception of third language contrasts. Hiromi Onishi (Univ. of Arizona, 503 Washington Ave. #4, Grinnell, Iowa 50112, honishi@email.arizona.edu)

The influence of L2 knowledge on the perception of L3 contrasts was examined in several experiments with two groups of Korean learners of Japanese. All participants have studied English as an L2 prior to beginning their study of Japanese as an L3. One group participated in a forced-choice identification experiment, and the other group participated in an AXB discrimination experiment with various Japanese contrasts. Additionally, both groups participated in a forced-choice English minimal pair identification experiment. Correlation between each group's performance in the Japanese experiment and the English experiment was examined in order to determine whether the perceptual level in English has any influence on the identification and discrimination of Japanese contrasts. The results of the correlation analysis suggested that the participants used increased knowledge in the L2 in a direct manner. That is, the better the participants performed on the L2 contrasts the better they also identified the L3 contrast, which is known to be difficult for them. During the L3 discrimination experiment, however, the participants seem to have used their increased L2 knowledge in a general manner. These results are considered an indication of L3 learners' general enhanced sensitivity as a result of the experience in L2 learning.

5pSC15. The acoustics of Mandarin tones in careful and conversational speech. Daniel Brenner (Univ. of Arizona, 814 E 9th St., Apt. 14, Tucson, AZ 85719-5322, dbrenner@email.arizona.edu)

A large proportion of the world's languages use phonological categories centering on vocal pitch to distinguish words. One of these, Mandarin, represents the largest native speaker population of any language in the world (SIL International, 2013). Although tones have long been foregrounded in phonetic/phonological work on Mandarin, and have been estimated to carry as much information in Mandarin phonology as vowels (Surendran and Levow, 2003), little is yet known about what happens to the tonal categories in conversation. This acoustic production study aims to detail the relationship between tones produced in casual conversation and those in the careful reading of a word list to determine the separability of tonal categories and the relative utility of acoustic cues in identifying those categories across speech styles. References: SIL International. (2013). "Statistical Summaries." In Lewis, M. Paul, Gary F. Simons, and Charles D. Fennig (eds.), *Ethnologue: Languages of the World*. Online resource: http://www.ethnologue.com/statistics/size. Surendran, Dinoj, and Gina-Anne Levow. (2003). "The Functional Load of Tone in Mandarin is as High as that of Vowels," in *Proceedings of the International Conference on Speech Prosody* 99–102.

5pSC16. Does first language prosodic transfer affect second language prosody? Charlotte F. Lomotey (Lit. and Lang., Texas A&M Univ., Commerce, 1818D Hunt St., Commerce, TX 75428, cefolatey@yahoo.com)

Learners of English have been found to transfer their L1 prosody into the prosody of L2 (Ramírez Verdugo, 2006). However, the effect of this transfer is not known or may not be universal. Besides, while English uses fundamental frequency in its intonation system, to indicate prominence in syllables and in phrases, and to signal differences in sentence intonation, Awutu uses it to signal lexical tone, a common phenomenon of tone languages. The present study investigates the effect of transfer of some prosodic features of Awutu, a language of Ghana, on English. To achieve this,

10 speakers of Awutu who are non-native speakers of English were asked to read narrow and broad focus statements and questions in both Awutu and English. The data were subjected to acoustic analysis for fundamental frequency using the Computerized Speech Laboratory. Preliminary findings show that Awutu speakers of English raise their fundamental frequency on focused words to show prominence. However, the pitch tracks of both statements and questions show that even though these speakers transfer some sentence prosody from Awutu, they do not show any consistency in the transfer. These findings suggest that the nature of L1 prosodic transfer into L2 may be language-specific.

5pSC17. The adaptation of tones in a language with registers: A case study of Thai loanwords in Mon. Alif Silpachai (Linguist., Univ. of California, Los Angles, 3170 Aintree Ln., Apt./Ste., Los Angeles, CA 90023, silpacha@usc.edu)

How do tones get adapted into languages with registers? This study examined loanword adaptation in which a language with registers borrows words from a language with lexical tones. In particular, this study presents an acoustic analysis of Thai loanwords in Mon, a language with two registers—one with tense voice and high f0 and the other with lax voice and low f0 accompanied by breathy voice. To investigate phonetic realizations, eight Mon speakers native to Thailand were recorded uttering 135 Thai loanwords in a carrier sentence. Results show that the tones in Thai loanwords get adapted as four level tones in Mon. In particular, loanwords with high tone have the highest f0, loanwords with mid tone have the second highest f0, loanwords with low tone and rising tone have the third highest f0, and loanwords with falling tone have the lowest f0. It is puzzling why Thai falling tone—not low tone—gets adapted as the lowest f0 in Mon. Results suggest that Mon spoken in Thailand may be developing lexical tones due to language contact.

5pSC18. Realization of Thai tone change in the Northern Thai dialect of Chiang Mai. Maureen Hoffmann (Dept. of Linguist., Univ. of Arizona, P.O. Box 210025, Tucson, AZ 85721, mhoffm@email.arizona.edu)

Recent studies have found evidence of tone change in progress among Thai speakers. In particular, changes in the high tone, traditionally considered a level tone, have caused some to suggest it should instead be considered a contour tone (Zsiga, 2008; Teeranon and Rungrojsuwan, 2009). However, the previous research has focused primarily on the Central Thai dialect found in Bangkok, the standard dialect of Thai. This study examines the current state of tones in the Northern Thai dialect of Chiang Mai, which has six contrastive tones, rather than the five found in Central Thai. This data allows for a comparison to both the Central Thai literature as well as previous studies of Northern Thai, to examine whether Northern Thai is undergoing tone change as well and whether it exhibits similar changes to those reported for Central Thai. Significant exposure to Central Thai via mass media as well as the education system, and widespread bi-dialectalism, may carry the influences of Central Thai tone changes into Northern Thai as well. This study aims to provide further insight into the ongoing changes in the Thai tonal space, in order to clarify the nature of Thai tones today.

5pSC19. The influence of lexical factors on word recognition by native English speakers and Japanese speakers acquiring English: An interim report. Kiyoko Yoneyama (English, Daito Bunka Univ., 1-9-1 Takashimadaira, Itabashi-ku, Tokyo, 175-8571, Japan, yoneyama@ic.diato.ac.jp) and Benjamin Munson (Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

In our earlier work [Yoneyama and Munson, J. Phonet. Soc. Jpn. 14-1 (2010)], we investigated whether neighborhood density and word frequency affect spoken word recognition in English by beginning and advanced Japanese L2 English speakers, and by native English speakers. Our study was modeled after the work of Imai *et al.* [J. Acoust. Soc. Am. (2005)]. The results indicated that there were strong effects of frequency and neighborhood density on the performance of all three groups of listeners. However, there was no clear evidence for an emerging "neighborhood competition" effect in the Japanese learners of English, contrary to Imai *et al.* Here we report two additional analyses of these data. The first uses log-linear modeling (i.e., the j-factors in Boothroyd and Nittrouer [J. Acoust. Soc.Am. (1998)]) to examine word recognition in the two groups. The second examines the influence of lexical variables on spoken word recognition response times in L1 and L2 speakers. Preliminary results suggest that the effect of word frequency and neighborhood density on these measures is similar for L1 and L2 speakers of English.

5pSC20. Effects of native language and speech rate on perceptual and decisional processing of voicing and syllable affiliation in stops. Noah H. Silbert (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., 344 French East Bldg., Cincinnati, OH 45267, noahsilbert@gmail.com), Kenneth de Jong (Linguist, Indiana Univ., Bloomington, IN), Byung-jin Lim (East Asian Lang. & Lit., Univ. of Wisconsin, Madison, WI), and Kyoko Nagao (Ctr. for Pediatric Auditory and Speech Sci., Nemours Biomedical Res., Wilmington, DE)

Previous work shows that variation in speech rate influences the perception of voicing distinctions (/b/-/p/) and syllable affiliation ("pea"-"eep"), and it is well-documented that native language influences how listeners perceive phonological distinctions. We analyze the influences of speech rate and native language in the perception of voicing and syllable affiliation by applying a model of perception and response selection to data from Japanese, English, and Korean listeners who identified the voicing and the syllable affiliation of (English) stops produced at slow, moderate, and fast rates. The fitted model indicates that for all three native language groups, perceptual salience decreases substantially as speech rate increases for both voicing and syllable affiliation. Even at slow rates, however, the salience of voicing is lower for coda than for onset stops. In addition, as rate increases, all three groups exhibit an increasing bias toward "onset" responses, with a bias toward "voiced" responses for coda stimuli and toward "voiceless" responses for onset stimuli. Despite broad similarities across all three groups, fine-grained patterns of perceptual salience and response bias vary with listeners' native language. These data and fitted models illustrate the utility of rate-varied speech in investigations of native language effects in speech perception.

5pSC21. Training effect on the second language learning for young learners using computer-assisted language learning system: Quantitative consideration on relationship among speech perception of the second language, learning experience and amounts of learning. Yuko Ikuma (English Education, Osaka Kyoiku Univ., 4-1-1-801, Bingo-cho, Nada-ward, Kobe, Hyogo 657-0037, Japan, yyikuma@mve.biglobe.ne.jp)

Longitudinal training experiment was conducted in order to examine the relation between the perceptual ability of English as a foreign language and amount of learning experiences beyond schools targeting Japanese elementary school students. Over four hundred students among the 3rd grade through 6th grade participated in this study. Three hundred and thirty-two students of them had learning experience beyond school, and the other, 134 students, did not. Students spent approximately 10 h of individualized computer-based training that focused on intensive auditory input. The result of t-test showed that the scores of the group of students who have previous learning experience exceeded the scores of the students in the other group at the beginning; however, at the end of the period, it revealed from the result of ANOVA that students without learning experience before starting learning English at school improved their sensitivity on perception of English syllable and some phonemes much more than the experienced. These results suggest that the appropriate perception training utilizing the auditory input is effective in cultivation of aural comprehension. Implications for foreign language education for young learners will be discussed. [Work supported by JSPS KAKENHI Grant-in-Aid for Young Scientists (B) 23730832, Japan.]

5pSC22. Intonational transfers in second language English speakers. Sergio Robles-Puente (Linguist., Univ. of Southern California, 3601 Watt Way; Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, roblespu@usc.edu)

Previous research on Spanish imperatives has demonstrated that their phonetic characteristics may not differ from those of declaratives. However,

under the right conditions, imperatives can be produced with up-stepped patterns where nuclear pitch-accents show higher F0 values than pre-nuclear ones. These circumflex configurations are never attested in declaratives (Robles-Puente, 2011). The current study concentrates on the imperatives and declaratives produced by 31 Mexican Spanish/English bilinguals and reveals that this variety of Spanish, unlike Iberian Spanish and English, allows not only imperatives but also declaratives to be produced without the aforesaid intonational constraint. Additionally, the English productions of the same speakers show circumflex configurations indicating a clear prosodic transfer characteristic of their mother tongue. Robles-Puente, Sergio. 2011, "Looking for the Spanish imperative intonation," in *Selected Proceedings of the 5th Conference on Laboratory Approaches to Romance Phonology*, edited by S. Alvord, pp. 153–164. Somerville, MA: CPP.

5pSC23. The effects of dialectal differences on the identification of English vowels by native and nonnative listeners. Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp)

This study attempts to investigate how dialectal differences of English affect the identification of English vowels by native and nonnative speakers of English. Served as listeners were native speakers of New Zealand English and Japanese. They heard and identified /i, ɪ, eɪ, ɛ, æ, ɑ̃, ʌ/ uttered by native speakers of New Zealand and American English. Repeated-measures ANOVAs were performed, respectively, for each listener group. The results revealed that there was no significant main effect of dialect ($p = 0.013$), but a main effects of vowels was found significant ($p \prec 0.001$). An interaction between dialect x vowels was also significant ($p \prec 0.001$). Pairwise comparisons revealed that NZ listeners identified NZ English /ɪ/, /ɑ̃/, /ʌ/ better than AM English counterparts ($p \prec 0.05$), but they identified AM English /æ/, /ɛ/ better than NZ English counterparts ($p \prec 0.05$). Native Japanese listeners, on the other hand, identified AM English vowels significantly better than NZ English vowels ($p \prec 0.001$). Particularly, they identified /i, ɪ, ɛ, æ/ uttered by AM English talkers than those uttered by NZ English talkers. However, native Japanese listeners identified NZ English /ɑ̃/ better than American English counterpart($p \prec 0.05$).

5pSC24. Perception of voicing of English word-final consonants: A comparative study of English listeners and Korean listeners. Ji Yea Kim (English Lang. and Lit., Seoul National Univ., 1 Gwanak-ro, Gwanak-gu, Seoul 151-745, South Korea, jiyeakim@snu.ac.kr)

This study aims to investigate the perception of word-final consonant voicing. Preceding vowel duration is of interest in comparing the perception of 7 English listeners (EL) and 7 Korean listeners (KL). Each listener was required to listen to 104 stimuli randomly composed of English voiceless and voiced consonants (e.g., "picks" and "pigs") and to choose from two options what they heard for each of the stimuli. There were 2 types of stimuli: original and manipulated. To manipulate vowel duration, for example, the vowel in the originally voiceless stimulus "picks" was lengthened, whereas the vowel in the voiced stimulus "pigs" was shortened by using PRAAT. The results show that, in the original stimuli, both groups tend to perceive voicing accurately, but ELs are better than KLs. It is assumed that the lower percentage of KLs' perception is due to the fact that there is no voicing contrast in Korean. In the manipulated stimuli, however, both groups generally fail to perceive voicing, and the number of stimuli whose voicing was never perceived was greater for ELs than that for KLs. This clearly indicates that ELs rely more on the voicing of following consonants than they do on the preceding vowel length.

5pSC25. Perception of epenthetic vowels in English /s/-initial clusters by Spanish-speaking second language learners of English. Maria Teresa Martinez-Garcia and Annie Tremblay (Linguist. Dept., Univ. of Kansas, 3700 Clinton Parkway, Apt. 212, Lawrence, KS 66047, maria.martinezgarcia@ku.edu)

Second language learners' (L2ers') perception and production of consonant clusters is influenced by the syllable structure of the native language (L1). This study investigates whether the perception of epenthetic vowels is partially responsible for why Spanish speakers have difficulty producing /s/ + Consonant ("sC") clusters in English, and whether it affects word recognition in continuous speech. Spanish, German L2ers of English, and native English speakers completed: (i) an AXB task with (/ə/)sC-initial nonce words (e.g., [əsman]-[sman]); (ii) a word monitoring task with (/ə/)sC-initial words in semantically ambiguous sentences (e.g., I have lived in that (e)state for a long time); and (iii) a production task with the same sentences as in (i). L2ers also took a word-familiarity rating task and a cloze test to assess their proficiency. For (i) and (ii), accuracy rates were recorded, and response times were measured from target onset. For (iii), acoustic analyses showed whether the L2ers' productions of sC-initial words contained an epenthetic vowel. Preliminary results suggest that perception difficulties may be partially responsible for Spanish speakers' production and word-recognition difficulties with sC-clusters in English, but production data suggest that articulatory problems may also play an important role. Proficiency does not seem to help overcome this difficulty.

5pSC26. The perception of English and Thai fricatives and affricates by Thai learners. Rungpat -. Roengpitya (Dept. of English, Faculty of Liberal Arts, Mahidol Univ., Thailand, 240 Soi 17, Rama IX Rd., Bangkok 10320, Thailand, rungpat@gmail.com)

English has eight voiceless-voiced fricatives /f, v, θ, ð, s, z, ʃ, and ʒ/ and two affricates /tʃ and dʒ/ in all positions. Thai, however, carries only two initial voiceless fricatives /f, s/ and one initial voiceless affricate /tʃ/. In the literature, the acoustic cues for fricatives include the frication noise, the amplitude, and the fundamental and formant frequencies on the adjacent vowels. This research explores how Thai listeners can perceive the English fricatives and affricates, as opposed to the Thai set. Thirty-one English and fifteen Thai words with fricatives and affricates were chosen. Two native-American male speakers read all English words, and a native-Thai female speaker read all Thai words. All the words were acoustically measured for the acoustic cues and digitally modified for all 312 tokens with different quality and quantity. Twenty native-Thai listeners (14 females and 6 males) listened and identified each token whether it contained which original fricative or affricate. The results revealed that the correct responses of the Thai learners were at a higher rate (90–100%) for the Thai original and modified tokens, and at a lower rate (30–100%) for the English set. It is hoped that this study will shed light on to future perceptual studies.

5pSC27. Effects of listener characteristics on foreign-accentedness rating of a non-standard English dialect. Andrea Morales and Natasha Warner (The Univ. of Arizona, 5242 S Hampton Roads Dr., Tucson, AZ 85756, andreamorales@email.arizona.edu)

This project analyzes what characteristics of listeners affect whether they perceive Chicano English as foreign-accented English. Many Americans assume Chicano English (CE) is non-native English spoken by native Spanish speakers, but CE is often spoken as a native dialect of English. CE is a very common dialect in Tucson, Arizona, and this project examines the correlation between listeners' ethnicity, familiarity with Hispanic people, and political stance on immigration, and their perception of CE as foreign-accented. Stimuli are sentences read by CE and other Tucson speakers that contain phonetic environments where CE has features that distinguish it from Standard American English (SAE). The listener population is Southern Arizonans of various ethnicities with varying degrees of exposure to CE and Spanish. The experiment uses a Foreign Accentedness Rating (FAR) task, as well as classification of stimuli as spoken by a Hispanic vs Anglo speaker and background questions on listeners' language background and political opinions. Highly accurate identification of ethnicity is predicted, as well as correlations between some measures of the listeners' background and strength of FAR rating of CE speakers. Conclusions involve the effect of long-term exposure to a local dialect and sociolinguistic status on perceived degree of foreign accent

5pSC28. Effect of native Mandarin dialects on English learners' use of prosodic cues to stress. Zhen Qin and Annie Tremblay (Dept. of Linguist, Univ. of Kansas, Blake Hall, Rm. 427, 1541 Lilac Ln., Lawrence, KS 66045, qinzhenquentin2@ku.edu)

Second-language learners (L2ers) weight phonetic cues to stress as a function of how these cues are used in the native language. This study investigates the effect of native dialects on the use of prosodic cues (F0 and duration) to English stress by native speakers (NSs) of Standard Mandarin (SM), Taiwanese Mandarin (TM), and English. Both TM and SM use F0 to realize lexical tones, but only SM uses duration to realize lexically contrastive stressed-unstressed vs stressed-stressed words. English NSs and intermediate-to-advanced TM-speaking or SM-speaking L2ers of English (at the same English proficiency) completed two sequence-recall tasks. In each trial, they heard four English non-words with trochaic and iambic stress, and pressed "1" and "2" to recall them in the correct order. In Experiment 1, participants heard natural stimuli (converging F0 and duration cues); in Experiment 2, the stress stimuli were resynthesized to contain only F0 cues, only duration cues, converging F0 and duration cues, or conflicting F0 and duration cues. In Experiment 1, all groups used naturally produced stress to recall English non-words. In Experiment 2, SM-speaking L2ers used duration more than TM-speaking L2ers to recall English non-words. Native dialect is suggested to be considered in L2 speech processing models.

5pSC29. Familiarity with a foreign accent aids perceptual accent adaptation. Cynthia P. Blanco (Linguist., Univ. of Texas at Austin, 113 East Hillside Dr., Greenville, South Carolina 29609, cindyblanco@utexas.edu), Emily Tagtow, and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

A change in speaker accent is reported to temporarily slow speech processing (Bradlow and Bent, 2003; Clarke and Garrett. 2004). Recent work suggests that this delay may be an artifact of task expectations and reflects a surprise effect, not the time needed for accent adaptation (Floccia *et al*., 2009). The present study tested listeners with high and low exposure to Spanish- and Korean-accented English to determine if frequent exposure to these accents decreases the surprise effect in an experimental setting. Participants listened to four blocks of meaningful sentences and responded to probe words; they heard a native-accented speaker in the first block and either native-, Spanish- or Korean-accented speakers in blocks 2 and 3. Results thus far show that the change from native-accented to foreign-accented speaker (block 1 to block 2) elicited a processing delay for participants in the Korean-accented condition, but not in the Spanish-accented condition. This pattern remained, but was somewhat attenuated, in the change from block 2 to block 3, when voice but not accent changed. These results show that extensive experience with a particular foreign accent (Spanish) outside the lab results in a smaller processing cost when listening to accented speech in the lab.

5pSC30. Predicting code-switches from phonetic information: The discourse marker *like* in Spanish-English code-switching. Page E. Piccinini (Linguist., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0108, ppiccinini@ucsd.edu)

The present study investigated whether Spanish-English bilinguals (L1 Spanish, English dominant) use phonetic cues to anticipate code-switches. Listeners were presented with four sets of 10 utterances. In a given set all utterances began in English or Spanish. All utterances included the discourse marker *like*. In each set, half of the utterances continued in the same language after *like* and half switched languages after *like*. Listeners only heard up to and including *like*. Listeners evenly sorted the utterances into two columns, "continues in English" or "continues in Spanish," to indicate which five utterances involved code-switching. Half of listeners received instructions in English and half in Spanish. Both sets of listeners were significantly above chance for stimuli beginning in English [$p < 0.05$]. Listeners who received Spanish instructions were also trending above chance for stimuli beginning in Spanish [$p = 0.08$]. This suggests listeners can use phonetic cues to anticipate switches from their dominant to their non-dominant language. Additionally, when language mode is the non-dominant language, listeners can also anticipate switches from their non-dominant to their dominant language. These results support a theory where both languages are somewhat activated at all times, allowing bilinguals to use phonetic cues to anticipate language switches.

5pSC31. Perception of English narrow and broad focus by native speakers of Mandarin Chinese. Ratree Wayland and Chelsea Guerra (Linguist, Univ. of Florida, 2801 SW 81st St., Gainesville, FL 32608, ratree@ufl.edu)

The aim of this study is to examine the ability to accurately perceive and comprehend English intonation patterns among native Mandarin speakers. Intonation patterns are patterns of rising and falling in pitch over the course of a full utterance. Both English and Mandarin make use of intonation patterns. However, unlike English, Mandarin is a tonal language in which pitch changes served to distinguish word meaning. The tonal patterns of words thus cause additional pitch fluctuation in the overall intonation of a Mandarin sentence. Sixteen Mandarin and 12 English speakers participated in the study. In the first task, participants were asked to listen to English sentences with either a falling or a rising intonation, and to decide whether the sentence is complete or incomplete. Participants' comprehension of English sentences produced with an intonation pattern focused on the verb, the noun or the entire sentence was examined. The results obtained indicated that (a) native speakers of English outperformed native Mandarin speakers on both tasks, that (b) both groups performed better on the second task, and that (c) the difference between the two tasks was greater among Mandarin speakers than among English speakers.

5pSC32. Prosodic profile of American Aviation English. Julia Trippe and Eric Pederson (Dept. of Linguist., Univ. of Oregon, Eugene, OR 97403-1290, trippe@uoregon.edu)

Aviation English (AE) is under scrutiny due to miscommunication between international pilots and controllers. To enhance public safety, since 2011, aviation professionals must prove technical and practical English proficiency. Previous studies measure AE speech accuracy by task performance and repeated elements (Barshi and Healy, 2011), and speech comprehensibility using native speaker judgments (Farris *et al*., 2008). The current study develops a quantifiable index for evaluating AE production based on prosody. Reasonably fluent prosody is critical to language comprehensibility generally, but since AE has no predictable intonation due to signal limitations, lack of function words, standard phraseology and rapid speech rate, we are specifically developing a rhythm profile of Native Speaker AE (NSAE) to evaluate Non-native Speaker AE production and model training methods for different first language (L1) prosodic types. We are training a speech aligner on tapes of US controllers to calculate a baseline for American NSAE. Our index will be generated using known metrics such as delta-V/C, %V (Ramus, 2000), PVI (Low *et al*., 2000), and varcoV/C (Dellwo, 2006). Since AE is derived from "stress-timed" English to be standardized and predictable, we predict that AE will exhibit a rhythmic signature comparable not only to English but to "syllable-timed" languages.

5pSC33. White-matter microstructure differs in adult bilingual and monolingual brains. Patricia K. Kuhl, Todd L. Richards, Jeff Stevenson, Dilara D. Can, Liv Wroblewski, Melanie S. Fish, and Julia Mizrahi (Inst. for Learning & Brain Sci., Univ. of Washington, Box 357920, Seattle, WA 98195, pkkuhl@u.washington.edu)

Behavioral research indicates that bilingual children and adults outperform monolinguals at executive function tasks, especially those related to cognitive flexibility, suggesting that experience with two languages alters brain structure. We investigated white-matter microstructure using Tract-Based Spatial Statistics (TBSS) in monolingual ($n = 15$) and Spanish-English bilingual ($n = 16$) adults, quantifying fiber tract organization in measures of directionality (fractional anisotropy, FA) and diffusivity perpendicular to the main axonal direction (radial diffusivity, RD). FA was significantly higher for monolinguals ($p < 0.05$, corrected) in three brain regions: the right posterior limb of the internal capsule, the right sagittal stratum that includes inferior frontal occipital fasciculus, and the right thalamus. RD was greater for bilinguals ($p < 0.05$, corrected) in multiple brain areas, most prominently in the cerebellum, inferior frontal occipital fasciculus, and superior longitudinal fasciculus. We interpret these differences in brain structure between monolinguals and bilinguals as consistent with the idea that bilingual language experience leads to a pattern of more diffuse connectivity in the brain, which may be related to increased cognitive flexibility skills.

5pSC34. Comparison of perceptual training and production training on tone identification. Shuang Lu, Ratree Wayland, and Edith Kaan (, Dept. of Linguist., Univ. of Florida, Turlington Hall 4131/P.O. Box 115454, Gainesville, FL 32611-5454, shuanglu@ufl.edu)

Previous studies have shown that short-term perceptual and production training can improve the comprehension and production of lexical tones by non-tone language speakers (e.g., Wang *et al.*, 1999; Leather, 1990). The current study compared the effectiveness of an identification-only training and an identification-plus-imitation training on lexical tone perception. Stimuli consisted of 12 monosyllables associated with three linear tones that resemble the level, rising and falling tones in Mandarin Chinese. Twenty participants first did a baseline identification task, and then received either an identification-only or an identification-plus-imitation training. The trainings were exactly the same except that the identification-plus-imitation training required participants to imitate the stimuli, while the identification-only training had participants utter the tone types of the stimuli (i.e., level, rising or falling). Lastly, all participants did the same baseline identification task again. The tone identification accuracy improved in both the identification-only and the identification-plus-imitation groups after training. Moreover, the identification-plus-imitation group identified the tones more quickly in the post-training task than in the pre-training task while the identification-only group did not show any improvement. These results indicated that the identification-plus-imitation training was more effective to improve the tone identification than the identification-only training.

FRIDAY AFTERNOON, 6 DECEMBER 2013 — CONTINENTAL 6, 1:00 P.M. TO 3:20 P.M.

Session 5pUW

Underwater Acoustics and Acoustical Oceanography: Sediment Acoustics: Modeling, Measurement, and Inversions III

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ARL, UT at Austin, 10000 Burnet Rd., Austin, TX 78758

Chair's Introduction—1:00

Invited Papers

1:05

5pUW1. Acoustic scattering from ocean sediment layers with multiple rough interfaces using finite elements. Marcia J. Isakson and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Acoustic scattering from the ocean bottom is a major component in shallow water reverberation and propagation as well as having a significant effect on the transmission of acoustic communication. However, boundary element models can only model scattering from a single rough interface. While some scattering models, such as the GeoAcoustic Bottom Interaction Model (GABIM), have considered scattering from layered sediments, these models are normally constrained to only one rough interface. Finite element models have been shown to accurately model scattering from both fluid and elastic boundaries, and, unlike conventional models based solely on the Helmholtz-Kirchhoff integral, are not limited to boundary interactions. In this study, a two-dimensional finite element model for scattering from two fluid layers and a fluid layer over an elastic layer is compared with perturbation theory and Kirchhoff approximation models to test the validity of considering the considering underlying interfaces flat. [Work sponsored by ONR, Ocean Acoustics.]

1:25

5pUW2. Adding thermal and granularity effects to the effective density fluid model. Kevin Williams (Appl. Phys. Lab. - Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, williams@apl.washington.edu)

Previously, an effective density fluid model (EDFM) was developed for unconsolidated granular sediments and applied to sand. The model is a simplification of the full Biot porous media model. Here two additional effects are added to the EDFM model: heat transfer between the liquid and solid at low frequencies and the granularity of the medium at high frequencies. The frequency range studied is 100 Hz–1 MHz. The analytical sound speed and attenuation expressions obtained have no free parameters. The resulting model is compared to ocean data.

1:45

5pUW3. Hybrid geoacoustic inversion scheme with an equivalent seabed model. Zhenglin Li and Renhe Zhang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21 Beisihuan West Rd., Beijing 100190, China, lzhl@mail.ioa.ac.cn)

Acoustic propagation in shallow water is greatly influenced by the properties of the bottom. The purpose of geoacoustic inversion is estimation of ocean bottom acoustic parameters such as sediment sound speeds, densities, and attenuations from measured acoustic fields. Especially, geoacoustic inversion could give low frequency attenuation, which cannot be measured by coring the sediment. Therefore, it has been paid much attention in recent years. A hybrid geoacoustic inversion scheme, which combines several inversion methods together to invert for the bottom parameters, has been proposed based on the fact that the bottom acoustic parameters have different sensitivities to the different physical parameters of acoustic field. This inversion scheme could avoid the problem of the multiple solutions, which are often companied with some geoacoustic inversion methods. The validity of the inversion scheme is verified in a series of sea experiments at different sites. In the experiment, six different sediment types: Fine Sand, Silty Sand, Sand Silty, Sand-Silty-Clay, Silty Clay and Mud, are included in an area in the Yellow Sea. The inverted bottom parameters could distinguish the atlas marked bottom type quite well. [Work supported by the National Natural Science Foundation of China under Grant Nos. 11074269 and 10734100.]

Contributed Papers

2:05

5pUW4. *In situ* measurements of sediment sound speed in the frequency band of 2–10 kHz at target and reverberation experiment site. Jie Yang and Dajun Tang (Acoust. Dept., APL-UW, 1013 NE 40th St., Seattle, WA 98105, jieyang@apl.washington.edu)

As part of the environmental measurements for TREX13 (Target and Reverberation EXperiment 2013), in situ measurements of surfacial sediment sound speed were carried out off Panama City, Florida, using a system called Sediment Acoustic-speed Measurement System (SAMS). SAMS consists of ten fixed sources positioned just above the seafloor, and one receiver which is driven into the seabed to a known depth. During TREX13, 10 deployments were made along the main reverberation track which is about 7.5 km in length. All measurements were made at a penetration depth of 3 m between 2 to 50 kHz, focusing on 2–10 kHz. Preliminary sediment sound speed results show variation from low sound speeds (muddy sites) to high sound speeds (sandy sites). A 3–5% of dispersion was observed at coarse sandy sites between 2 and 10 kHz, whereas little dispersion was observed at muddy sites. [Work supported by ONR.]

2:20

5pUW5. Assessing grain size as a predictor of mid-frequency bottom backscattering strengths. Roger C. Gauss, Edward L. Kunz, and Altan Turgut (Acoust. Div., Naval Res. Lab., Code 7164, 4555 Overlook Ave., S.W., Washington, DC 20375-5350, roger.gauss@nrl.navy.mil)

Scattering from the seabed can be a complex mix of surface roughness and volume heterogeneity contributions. A series of mid-frequency (MF; 1.5–4.5 kHz) bottom backscattering strength data collected by the Naval Research Laboratory at a number of shallow-water locations (Stanton Banks, Malta Plateau, Heceta Bank) is first used to demonstrate the inadequacies of using Lambert's Law to model bottom backscattering strengths, and that more general empirical power laws, where not only the strength but the angular exponent can vary, are needed to match the data at a given frequency. The Stanton Banks data, where sediment types range from mud to gravel, are then used to explore the extent to which easy-to-access geophysical data (such as surficial grain size distributions from bottom grab samples) may be capable of providing suitable estimates of key model inputs (such as sediment sound speeds/attenuations, density and roughness/volume spectral strengths/exponents). These results show that both grain size and "bottom type" are in general unreliable predictors of the measured MF bottom backscattering strengths, and that a physics-based modeling approach coupled with in-situ environmental characterization is required. [Work supported by ONR.]

2:35

5pUW6. Estimates of sediment volume heterogeneity spectra from several distinct shallow water locations. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu)

Theory indicates that sediment volume heterogeneities tend to dominate seabed scattering above the critical angle. However, recent evidence indicates that scattering from volume heterogeneities can also be the dominant mechanism below the critical angle. This raises questions about the nature and scales of volume heterogeneities in marine sediments. Direct measurements of sediment heterogeneities have been performed on cores using for example x-ray CT scans, however this and related methods only sample a very small volume. In this paper, a method is presented for estimating sediment heterogeneity spectra from acoustic reverberation data where the sediment volume probed is order 10^7 m^3. The large averaging volume permits measuring a wide range of spatial frequencies and tends to emphasize persistent scales. Resulting sediment volume spectra from several different shallow water regions will be presented. [Research sponsored by the ONR Ocean Acoustics.]

2:50

5pUW7. Laboratory measurements of sound speed and attenuation in sandy sediments. Yi-Wang Huang, Shi-e Yang, Qi Li, Sheng-Qi Yu, Fei Wang (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China, huangyiwang@hrbeu.edu.cn), Dajun Tang, and Eric I. Thorsos (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Marine sediments exist universally as the lower boundary for sound propagation in ocean waveguides, and knowledge of the properties of these sediments is important for accurate modeling of sound propagation and reverberation. In order to test theory predictions of the frequency dependence of sound speed and attenuation, it is necessary to have accurate information on the sediment properties, which is most easily done in a laboratory environment. Initial results reported here were done at high frequency in a small tank, as a preliminary step before making similar low frequency measurements in a much larger tank. A sandy sediment was used and the sound speed and attenuation were measured through different thicknesses of the sample. In the frequency range of 90–170 kHz, the measured sound speed was 1757–1767 m/s, and the attenuation was 22–30 dB/m. The sound speed dispersion was found to be very weak, as expected, and much smaller than the measurement uncertainty. The attenuation was found to increase approximately linearly with frequency. The measured sound speed agrees well with Biot theory predictions, while the measured attenuation is higher than Biot predictions, most likely because the measurement include effects such as volume scattering not taken into account in the theory.

3:05

5pUW8. Comparison of the finite element method with perturbation theory and the small-slope approximation for acoustic scattering from one-dimensional rough poroelastic surfaces. Anthony L. Bonomo, Marcia J. Isakson, and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

The finite element method is used to address the problem of acoustic scattering from one-dimensional rough poroelastic surfaces. The poroelastic sediment is modeled following the Biot-Stoll formulation. The rough surfaces are generated using a modified power law spectrum. Both backscattering strengths and bistatic scattering strengths are calculated. These results are compared with lowest order perturbation theory and the lowest order small-slope approximation, as extended to the case of scattering from poroelastic surfaces. It is known that these approximate methods are sufficient for the study of rough surface scattering in the case of sediments modeled as fluids. This work seeks to assess whether or not these methods are accurate when applied to the case of poroelastic sediments. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]