Session 5aBA

Biomedical Acoustics: Medical Ultrasound

Siddartha Sikdar, Chair
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Contributed Papers

8:00

5aBA1. Numerical evaluation of absorbing boundary layers for the transient Khokhlov–Zabolotskaya–Kuznetsov Equation. Xiaofeng Zhao and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, zhaoxia6@msu.edu)

FOCUS, the "Fast Object-Oriented C++ Ultrasound Simulator" (http://www.egr.msu.edu/~fultras-web), simulates nonlinear ultrasound propagation in the time-domain by numerically evaluating the transient Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation. KZK simulations in FOCUS previously required a computational grid with a large radial distance relative to the aperture radius to reduce the effect of reflections from the boundary. To decrease the size of the grid required for these calculations, an absorbing boundary layer derived from the power law wave equation provides an alternative to the stretched coordinate system in a perfectly matched layer. Simulations of the linear pressure fields generated by a spherically focused transducer with an aperture radius of 1.5 cm and a radius of curvature of 6 cm are evaluated for a short pulse with a center frequency of 1 MHz and a peak surface pressure of 0.5 MPa. Numerical results for linear KZK simulations with and without the absorbing boundary layer are compared to an on-axis analytical solution. Results of nonlinear KZK simulations with and without the PML are also evaluated, and all of these show that the absorbing layer effectively attenuates the wavefronts that reach the boundary of the computational grid. [This work was supported in part by NIH Grant R01 EB012079.]

8:15

5aBA2. An improved interpolation approach for rapid simulations of pulse-echo ultrasound imaging. Leslie P. Thomas and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, lt.thomas.jr@gmail.com)

New B-mode image simulation routines in FOCUS (http://www.egr.msu.edu/~fultras-web) apply linear interpolation to a finely sampled pressure signal that is calculated once per scatterer and then reused for different time delay and amplitude values in subsequent A-lines. This approach achieves more accurate results in less time than simulations that calculate the pressure waveforms on a coarser grid followed by cubic spline interpolation. To demonstrate this result, simulations are performed using 24 element sub-apertures in a linear array consisting of 192 rectangular elements that are 5 mm high and 0.5133 mm wide with 0.1 mm center-to-center spacing. The center frequency of the excitation is 3 MHz for simulations of a computer phantom with 100,000 scatterers. Results for each approach are compared to a simulated reference signal calculated with the impulse response for a sampling frequency of 1 GHz. To achieve an RMS error of 6%, FOCUS requires linearly interpolated signals sampled at 16 MHz, where the stored signal was sampled at 64 MHz. The total computation time for the linear interpolation approach was 36 min, whereas the cubic spline interpolations required a frequency of 20 MHz and took 49 minto run to achieve a similar RMS error. [This work was supported in part by NIH Grant R01 EB012079.]

8:30

5aBA3. Toward monodisperse ultrasound-triggered phase-shift emulsions using differential centrifugation. Kyle Stewart (Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, Cardiovascular Ctr., Rm. 3972, Cincinnati, OH 45267-0586, stewake@mail.uc.edu), Kirthi Radhakrishnan, and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Acoustic droplet vaporization (ADV) is a process that enables the *in situ* production of microbubbles from an injected perfluorocarbon emulsion and it has been investigated for imaging and therapeutic applications. High-speed mechanical shaking rapidly produces a polydisperse emulsion (~10^{10} droplets/mL) with droplets ranging from less than 400 nm to greater than 15 μm. The ADV pressure amplitude threshold is higher and fraction of transitioned droplets lower for droplets smaller than approximately 2 μm in diameter. Droplets greater than approximately 8 μm in diameter are not suitable for systemic administration. Therefore, high-speed mechanical shaking produces many droplets of limited utility. Differential centrifugation has been used as a size isolation technique for polydisperse ultrasound contrast agents. By applying a similar technique to a perfluorocarbon emulsion, the volume-weighted fraction of droplets between 2 and 5 μm was increased from 29±2% to 93±3%. The transition efficiency of the droplets between 2 and 5 μm was the same regardless of whether a polydisperse or monodisperse distribution was insonfied with 2 MHz ultrasound. The ADV pressure threshold of differentially centrifuged droplets was similar to non-centrifuged droplets. [Supported in part by NIH grant KL2 TR000078.]

8:45

5aBA4. Characterizing the pressure field in a modified microbubble flow cytometer: Using a laser Doppler vibrometer to validate the internal pressure. Cheng-Hui Wang (Inst. of Appl. Acoust., Shaanxi Normal Univ., Xi'an, Shaanxi, China), Camilo Perez (BioEng. and Ctr. for Industrial and Medical Ultrasound - Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Appl. Phys. Lab. - CIMU, Seattle, WA 98105-6698, camipiri@uw.edu), Jarred Swalwell (Oceanogr., Univ. of Washington, Seattle, WA), Brian MacConaghy (Ctr. for Industrial and Medical Ultrasound - Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Juan Tu (Key Lab. of Modern Acoust., Nanjing Univ., Nanjing, China), and Thomas J. Matula (Oceanogr., Univ. of Washington, Seattle, WA)

Previously, a flow cytometer was modified with a PZT transducer in order to study the radial oscillations of statistically significant numbers of microbubbles (J. Acoust. Soc. Am. **126**, 2954–2962, (2009)). We reported the results of pressure calibration for transient sonication in a recent symposium (ASA, Indianapolis, 2013). Here, we report the results of pressure calibration for steady-state sonication. Because the flow channel width (< 200 μm) is too narrow to insert our hydrophones, we rely on finite element analysis (FEA) to predict the acoustic pressure field. In this study the simulation results were compared to Laser Doppler Vibrometer *in-situ* measurements of the velocities of the surface of the flow chamber. The OFV-534 sensor head with OFV-5000 controller (Polytec, Irvine CA) was mounted so that the laser reflected off the proximal outer surface of the flow chamber. The FEA model coupled structural vibration and linear acoustic physics to calculate the steady state pressure. The FEA model compared

favorably with the LDV measurements. The FEA simulations were used to predict the pressure field, leaving only the shell elasticity and viscosity χ and κ as unknown variables in the bubble dynamics model. Excellent fits to Optison bubble oscillations were obtained.

9:00

5aBA5. The features of sound propagation through human lungs, revealed by transmission sounding with phase manipulated acoustic signal of 80–1000 Hz frequency band. Vladimir Korenbaum and Anton Shiryaev (Pacific Oceanologic Inst., 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru)

The sound propagation in human lungs remains poor studied. We studied the phenomenon in a frequency range of 80–1000 Hz by means of sounding lungs with phase manipulated signal injected into mouth and right/left supraclavicular chest areas. The installation included the subwoofer (mouth) and the shaker (supraclavicular zone), both fed through the power amplifier from the sound card of laptop, and a system of accelerometer sensors, connected to PowerLab (ADInstruments). Transmitted signals were recorded above medial trachea and in eight medial/basal posterior chest positions of 11 volunteers. Distances emitter-sensor and sensor-sensor were measured by pelvimeter. Convolution technique was applied for signal processing. Two to three arrivals of the signal were measured above trachea for sounding through mouth. The second arrival is treated as a reflection through air lumen from distal bronchi level with the distance estimated as 23 ± 5 cm (sound speed 200 m/s). Four to five arrivals with various median velocities were measured in posterior chest positions for sounding through mouth as well as both supraclavicular chest areas. The air-structural (air—tissue) and pure structural (tissue) transmission paths are identified for sounding through mouth. Meanwhile only pure structural transmission paths are supposed for chest sounding. [This study was supported by RFBR grant 13-08-00010.]

9:15

5aBA6. Simultaneous measurement of sound pressure and temperature of tissue mimicking material by an optical fiber Bragg grating sensor. Keisuke Imade, Daisuke Koyama, and Iwaki Akiyama (Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe 610-0321, Japan, duo0315@mail4.doshisha.ac.jp)

An optical fiber Bragg grating (FBG) sensor is capable of simultaneous measurement of sound pressure and temperature. The FBG is a type of refractive index grating constructed in a short segment of optical single mode fiber that reflects particular wavelengths of light. This Bragg wavelength depends on both of the sound pressure and the temperature. We reported the simultaneous measurement of sound pressure and temperature in water under the conditions of medical application. In this study, temperature rise of a tissue mimicking material (TMM) caused by exposure to ultrasound in the range of 0.3 to 5 MPa were measured. The TMM is made of agar-gel with Glycerin solution of 11.21%. The FBG of 1 mm in length constructed in the optical fiber of 0.25 mm in diameter was used in the experiments. As a result, as the measured sound pressure was 5 MPa, the measured temperature rise was 6.0 degree. Thus this sensor is capable of simultaneous measurement of sound pressure and temperature rise of biological tissues exposed to the ultrasound. [This study was supported by MEXT-Supported Program for the Strategic Research Foundation at Private Universities, 2013–2017.]

9:30

5aBA7. On the use of local speckle field as a correction factor for shear modulus estimates based on multiple-track-locations methods. Laurentius O. Osapoetra (Phys. & Astronomy, Univ. of Rochester, 500 Joseph C Wilson Blvd. CPU Box 271443, Rochester, NY 14627, loscar@pas.rochester.edu), Jonathan H. Langdon, and Stephen A. McAleavey (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Acoustic-radiation-force impulse (ARFI) imaging for characterization of shear modulus of biological tissues employs either multiple-track-locations (MTL) methods or single-track-location (STL) methods. MTL estimates of shear modulus at different depths suffer from more variability compared to those of STL estimates. Our studies have shown a significant correlation between bias in shear-wave arrival time estimates and local speckle field lateral statistics. We propose using the local speckle field as a surrogate for the unknown bias apparent in MTL shear-wave speed estimates. This local speckle field is determined using a "swept-receive acquisition" that is produced by holding the transmit beam fixed while laterally translating the receive aperture. Application of various lateral weighting functions to the swept-receive image results in an approximate compensation to the tracking bias. In this study, we implement our technique on simulation and experimental data from gelatin phantoms. Additionally, the effect of varying transmit-receive aperture f/# on the accuracy of our compensation method is investigated in simulation. Finally, the quality of compensated MTL shear modulus images are quantified and compared to those of the uncompensated MTL and of STL methods in matched tissue mimicking phantom experiments.

9:45

5aBA8. Iterative reconstruction of the ultrasound attenuation coefficient from backscattered signals. Natalia Ilyina (Dept. of Cardiovascular Sci., KU Leuven, UZ Herestraat 49 - box 7003 50, Leuven, België 3000, Belgium, natalia.ilyina@uzleuven.be), Jeroen Hermans (DoseVue, Hasselt, Belgium), Erik Verboven, Koen Van Den Abeele (Dept. of Phys., KU Leuven Kulak, Kortrijk, Belgium), Emiliano D'Agostino (DoseVue, Hasselt, Belgium), and Jan D'hooge (Dept. of Cardiovascular Sci., KU Leuven, Leuven, Belgium)

Estimation of the local acoustic attenuation from backscattered signals has several clinical applications but remains an open problem. Most of the proposed solutions relate the observed spectral changes directly to the theoretical predictions. However, these methods make a number of approximations and require correction strategies for acoustic wave phenomena that are not accounted for in the model (e.g., non-linearity). In this study, attenuation was estimated by successively solving the forward wave propagation problem for different attenuation coefficients and by matching the calculated backscattered signals to the observed one. For the forward problem, the effects of attenuation, nonlinear distortion, reflection and scattering were taken into account. The proposed approach was validated on simulated data and data recorded in six tissue mimicking phantoms and was compared to conventional methods. The relative error of the attenuation coefficient remained below 10% for the simulated and phantom data. The conventional methods showed a comparable performance on the simulated data, but their error significantly increased in the phantom study. The proposed method outperformed state-of-the-art attenuation estimators. Moreover, it can be used to estimate local non-linearity. In future work, the propagation model will be extended to 3D and diffraction effects will be included.

10:00–10:15 Break

10:15

5aBA9. Non-invasive monitoring of Achille's tendon stiffness variations *in-vivo* using mechanical vibrations. Muhammad Salman (Systems and Mech. Eng., Southern Polytechnic State Univ., 1100 S Marietta Pkwy # G-172, Marietta, GA 30060, msalman@spsu.edu) and Karim Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

A non-invasive monitoring technique of laser Doppler vibrometer (LDV) is used to find the stiffness of Achilles tendon. It is difficult for an ultrasound to collect the elastographic images of high stiffness areas such as Achilles tendon. Magnetic resonance images (MRI) technique is expensive and requires extensive training of the clinicians for elastographic image processing. This non-invasive technique needs short setup time and a simple physical structure for the data collection. A shaker is used as an excitation source, which generates waves on the tendon surface. This dynamic elastography technique measure wave velocities by an LDV. Achilles tendon is excited from 10 Hz to 1000 Hz using shaker and sensed by the LDV at four positions, which are one cm apart. Cross correlation signal processing is used for finding the time delays of the waves approaching each sensor location. It is found that as the contraction level increases, tendon stiffness increases. A comparison of average and varied stiffness values is shown in Achilles tendon. This technique may assist clinicians in characterizing muscle tone changes due to sport injuries in tendon.

10:30

5aBA10. Empirical mode decomposition-discrete Hilbert transform based solution to detect decompression-induced gas bubble from Doppler ultrasound signal. Md Iqbal Aziz Khan and Takayoshi Nakai (Graduate School of Sci. and Technol., Shizuoka Univ., Hamamatsu, 3-5-1 Johoku, Naka-ku 432-8561, Japan, iqbal_aziz_khan@yahoo.com)

This paper concerns the detection of decompression-induced gas bubble from underwater construction workers Doppler ultrasound signal based on empirical mode decomposition (EMD), discrete Hilbert transform (DHT), and some parameters, which could be useful to detect gas bubble. EMD in combination with DHT is used to generate time-frequency-energy distribution and systolic phase is detected from that distribution. The properly detection of the systolic phase is the most important task to the detection of gas bubble. Time-varying mean frequency (TMF) is determined from instantaneous amplitude and instantaneous frequency. An algorithm is applied to extract some specific segments from TMF. Then the segments are considered as gas bubble on the basis of some parameters such as spreading in frequency, ratio of the gas bubble signal to the background signal, ratio of the gas bubble signal's total energy to the background signal, and rising rate.

10:45

5aBA11. Three-dimensional pulsation of rat carotid artery bifurcation observed using a high-resolution ultrasound imaging system. Changzhu Jin, Kweon-Ho Nam, and Dong-Guk Paeng (Ocean System Eng., Jeju National Univ., Jeju, Jeju Self Government Province 690-756, South Korea, yustchang@gmail.com)

The arterial structure experiences cyclic pulsation in three-dimension (3-D) by pulsatile blood flow. Here we reconstruct the 3-D carotid artery bifurcation (CAB) geometry from three rats to observe the pulsatile variation of the carotid artery geometry by a high-resolution ultrasound imaging system (Vevo 770, VisualSonics, Canada). Two experienced observers manually segmented the cross-sectional ultrasound images to outline the arterial lumen. From the constructed geometry of three subjects, we observed that the CAB geometry favorably expanded to anterior/ posterior direction which parallel to sagittal plane. Furthermore, an *in vitro* blood flow experiment using an anthropomorphic flow phantom was implemented to more clearly understand the asymmetric pulsation phenomenon. Finally we confirmed the elastic bifurcation geometry favorably expands in a direction of bifurcation, both *in vivo* and *in vitro* measurements, which derived by bifurcated flow. This finding, the asymmetrical pulsation phenomenon of carotid bifurcation in 3-D, may be useful in understanding hemodynamic etiology of cardiovascular diseases and also provide more realistic input data for computer simulation of hemodynamics. [This work was supported by MSIP Korea, under the C-ITRC support program (NIPA-2014-H0401-14-1002) supervised by the NIPA and also supported by the Richard Merkin Visiting Fellowship Program of the Focused Ultrasound Foundation.]

11:00

5aBA12. Ability of skeletal muscle to protect bones and joints from external impacts: Acoustical assessment. Armen Sarvazyan (Artann Labs., 1459 Lower Ferry Rd., Trenton, NJ 08618, armen@artannlabs.com), Sergey Tsyuryupa (Artann Labs., Lambertville, NJ), and Oleg Rudenko (Phys. Dept., Moscow State Univ., Moscow, Russian Federation)

One of the major (but the least studied!) functions of skeletal muscle is protecting the skeletal system from external impacts by absorbing and redistributing the energy of mechanical shock in time and space. During muscle contraction, its elasticity modulus is greatly increased, which partly unloads adjacent bones and skeletal joints. Muscle viscosity is also greatly increased helping to absorb and dissipate dangerous shocks. Elasticity and viscosity data may be obtained using the measurement of shear wave velocity and attenuation. In this study, we investigated changes in the velocity and attenuation of shear acoustic waves in an anisotropic tissue phantom mimicking skeletal muscle under different level of tension of the fibers imbedded in the phantom. Stretching the fibers simulates the muscle contraction. It is shown that both velocity and attenuation of shear waves propagating along the fibers significantly increase with fiber tension while they are negligibly affected in the case of wave propagation across the fibers. We developed a theory for propagation of shear waves in anisotropic medium simulating the muscle and muscle contraction. Equations for the speed and attenuation of various modes of shear acoustic waves are derived. Theoretical predictions are in agreement with experimental data. [NIH R21AR065024.]

11:15

5aBA13. Novel use of ultrasound imaging to decode activity of forearm muscles for upper extremity prosthetic control. Nima Akhlaghi (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, nakhlagh@gmu.edu), Mohamed Lahlou (BioEng., George Mason Univ., Fairfax, VA), Brian J. Monroe (Hanger Clinic, Fairfax, Virginia), Parag V. Chitnis (BioEng., George Mason Univ., Fairfax, VA), Huzefa Rangwala, Jana Kosecka (Comput. Sci., George Mason Univ., Fairfax, VA), Joseph J. Pancrazio, and Siddhartha Sikdar (BioEng., George Mason Univ., Fairfax, VA)

With the recent developments in the electro-mechanical design of upper extremity prosthetics, the need for more advanced control strategies for such prosthetics has increased. Current commercially available prostheses based on myoelectric control have limited functionality, which leads to many amputees abandoning use. Myoelectric control using surface electrodes has a number of limitations, such as low signal to noise ratio and lack of specificity for deep muscles. To address these limitations, and enable more intuitive dexterous control, we propose a new strategy for sensing muscle activity based on real-time ultrasound imaging. Ultrasound imaging of the forearm muscles was performed on six healthy volunteers and a transradial amputee using a Sonix RP system with a 5–14 MHz linear array transducer. Images were analyzed to map muscle activity based on the changes in the ultrasound echogenicity of the contracting muscles during different complex movements, and used to control a virtual prosthetic hand. Individual digit movement could be decoded with 97% accuracy, and 15 different complex grasps could be decoded with 87% accuracy. In a transradial amputee, we were able to differentiate between seven different movements. These preliminary results demonstrate the feasibility of using ultrasound imaging for control of upper extremity prostheses.

11:30

5aBA14. Validity of acoustic method for the assessment of whole-body hydration status. Alan Utter, Mason C. Calhoun, Steven R. McAnulty, Jeffrey M. McBride, Jennifer Zwetsloot, Melanie Austin, Jonathan D. Mehlhorn, Lesley Sommerfield (Health and Exercise Sci., Appalachian State Univ., 111 Rivers St., Boone, NC 28608, utterac@appstate.edu), Sergey Tsyuryupa (Artann Labs., Trenton, NJ), and Armen Sarvazyan (Artann Labs., Lambertville, NJ)

In almost any sport, athletes undergoing dehydration often suffer from numerous dehydration related injuries. The purpose of this study was to evaluate the validity of acoustic method to detect changes in the hydration status of athletes after undergoing acute dehydration and a 2-hour rehydration protocol. The acoustic method of assessing body hydration status is based on the experimental fact that ultrasound speed in muscle is a linear function of the tissue water content. The assessment of water imbalance was conducted by measuring speed of ultrasound in the calf muscles using through transmission method. Eighty-two male and female collegiate athletes were examined to detect changes in hydration status before and after undergoing 3% acute dehydration. Results demonstrated that the changes of ultrasound velocity are in average about 1.1 m/s per 1% of body dehydration and ultrasound velocity in muscle potentially may serve as a measure of body hydration status. However, ultrasound speed measurement using through transmission mode implemented in this study is highly dependent on the positioning of the probe: even slight variation in the acoustic path results in significant changes in the measured values, which may results in unacceptable error. A solution to this problem is proposed and discussed. [NIH2R44AG042990.]

11:45

5aBA15. Design and characterization of a sensitive optical micromachined ultrasound transducer. Suzanne M. Leinders (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CJ, Netherlands, s.m.leinders@tudelft.nl), Wouter J. Westerveld (Optics Res. Group, Delft Univ. of Technol., Delft, Netherlands), Jose Pozo, Paul van Neer (TNO, Tech. Sci., Delft, Netherlands), H. P. Urbach (Optics Res. Group, Delft Univ. of Technol., Delft, Netherlands), Nico de Jong, and Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Delft, Netherlands)

Novel 3D intravascular or transesophageal ultrasound approaches require transducer arrays containing many small elements. Conventional piezo-electric techniques face fabrication challenges due to narrow kerfs and dense wiring. Micro-machined alternatives like CMUTs and PMUTs lack either sensitivity or bandwidth to fully compete. Therefore we developed an opto-mechanical ultrasound sensor. The absence of wiring makes it MRI compatible. The developed OMUT contains integrated photonics, which is fabricated using standard silicon-on-insulator technology, providing a small footprint and enabling mass production and ease of integration. The sensor consists of a straight waveguide and a micro-ring resonator integrated on a 124 μm wide, 2.7 μm thick acoustical membrane. Light, passing the waveguide, is partly coupled into the ring resonator. A dip appears in the spectrum of the transmitted light at the resonance wavelength of the micro-ring. If the acoustical membrane and hence the micro-ring is deformed due to an incident ultrasound wave, this is observed as a shift in the resonance wavelength of the ring. This paper presents the construction and characterization of our device. Measurement results of the linearity, frequency response, sensitivity and temperature dependence are compared with a model. The results demonstrate that our OMUT is a promising basis for future ultrasound arrays.

FRIDAY MORNING, 22 MAY 2015 KINGS 4, 8:30 A.M. TO 11:50 A.M.

Session 5aMU

Musical Acoustics: Non-Western Musical Instruments and Performance Spaces

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

Chair's Introduction—8:30

Invited Papers

8:35

5aMU1. Understanding timbral effects of multi-resonator/generator systems of wind instruments in the context of western and non-western music. Jonas Braasch (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Wind instruments are often modeled as a coupled generator-resonator system, which in case of the clarinet consists of a reed and a cylindrical pipe. However, the fact that the throat and oral cavities behind the tone generator also play a critical role is often neglected. While the resonance pipe of a wind instrument can be adjusted in length through keyholes or valves, its width is fixed. The opposite can be observed for the human resonance system where the length is fixed but the dimensions of the cross-section are variable. This is important fact is routinely used to shape the timbral qualities of tones and to determine the tones' fundamental frequencies—for example, to generate high notes above the normal range on a saxophone. In many music cultures, time-varying timbral modifications are more important than melodic aspects, for example, in traditional digeridoo practice. A third resonator system is the room the musical instrument is performed in. This can also have important timbre shaping aspects, emphasizing, for example, the brilliant sound of Bach trumpets. An extended saxophone [Braasch, 2014, J. Acoust. Soc. Am. **135**, 2245] and convolution reverb is used to demonstrate and analyze the effects of multi-resonator/generator systems for this presentation.

8:55

5aMU2. "Good acoustics" is culturally determined: Evidence that prehistoric performance space selection was based on different world views. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com)

"Ideal" performance spaces are in the ear of the listener. Cases will be presented to argue that the response to a particular set of acoustic characteristics of a space is a function of the world view of the audience/performers. For example, in today's modern society that values speech intelligibility and appreciation of polyphonic classical and contemporary music, a distinct echo in a performance space is considered an unforgivable fault. However, in the ancient past when wave characteristics of sound were not understood, virtual sound effects arising from complex sound wave interactions (echoes, reverberations, interference patterns, etc.) were misinterpreted as invisible beings (echo spirits, thunder gods, sound absorbing bodies, etc.) as described in ancient myths around the world. Some of these myths contain accounts of purposeful searches for the best echoes, which were worshiped. Scientifically conducted experiments involving blindfolded participants show how various ambiguous sounds can be interpreted in more than one way (like optical illusions). Sound level measurements demonstrate that cave paintings and canyon petroglyphs were placed in locations with the strongest echoes. Prehistoric flutes have been found in reverberating caves. These experiments thus can help in understanding our ancestors' perceptions and reactions to sounds they considered supernatural (https://sites.google.com/site/rockartacoustics/).

9:15

5aMU3. The origins of building acoustics for theater and music performances. John Mourjopoulos (Audio & Acoust. Technol. Group, Elec. and Comput. Eng., Univ. of Patras, Audio Group, WCL, Electricak and Comput. Eng. Dept. University of Patraser E, Patras 26500, Greece, mourjop@upatras.gr)

The ancient Greek amphitheaters and roofed odeia represent the earliest examples of acoustics utilized for enhancing theatrical and music performances over large public audiences, often up to 15,000 participants. Such an early achievement, more than 2000 years ago, was possibly crucial for the foundation of these performance-based art forms within the ancient Greek society and the eventual geographical spread of these buildings. Through the continuous evolution during the Roman era, via the Renaissance theaters and the modern concert halls, these specially constructed spaces allowed the performance-based art forms to be sustained and evolve as essential constituents of the western cultural heritage and civilization. We shall first consider some historical and architectural properties of these early performance spaces and then summarize their design principles and acoustic parameters. Acoustic properties were different for spaces used for theatrical and music performances: open-air theaters for drama were manly utilizing early reflections (e.g., up to 40 ms) to achieve perfect speech intelligibility for listener distances even beyond 60 m; in contrast, roofed-odeia had acoustics appropriate for music with prominent reverberation comparable to that of modern concert halls. The presentation will include auralization demos for important ancient theaters and odeia.

9:35

5aMU4. Conch-shells, bells, and gongs in Hindu temples. Marehalli Prasad (Mech. Eng., Stevens Inst. of Technol., 519 Hudson St., Carnegie Bldg., Hoboken, NJ 07030, mprasad@stevens.edu)

Acoustics plays an important role in Hinduism and Hindu temples and worship practices. Several instruments such as conch-shells, bells, and gongs due to their musical qualities are used to enhance the spiritual experience of devotees in Hindu temples. These instruments are used to augment the acoustically rich Vedic chanting and singing. These instruments are generally used in a room called *Ardha Mantapa* (in Sanskrit), which is coupled to the Sanctum Sanctorum (called as *Garbha Griha* in Sanskrit) in a Hindu temple. The *Garbha Griha* and the *Ardha Mantapa* play an important role in the acoustical environment in the temples. These instruments are used both individually as well as collectively during the worship. These instruments through their tonal quality complement the soundscape of the environment. This presentation deals with spectral measurements of these instruments and also the acoustical aspects of the temple space. Measurements taken in two Hindu temples will be presented.

9:55

5aMU5. On the acoustics of Maya pyramids. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514, dlubman@dlacoustics.com)

On my first visit to the ancient Maya ceremonial city of Chichen Itza in 1998, I discovered an amazing sonic phenomenon at a limestone pyramid known as the temple of Kukulkan. This four-sided 30 m high pyramid resides in a large open plaza with no nearby structures to produce echoes. But clap your hands at the pyramid and you hear a brief chirped echo. This was unexpected. Normally, echoes are delayed replica of their stimuli (a single handclap in this case). Yet this echo sounds unlike the handclap. It is a downward-gliding and harmonic-rich chirp. I recorded the echo for later study. Surprises continued when sonograms of the chirped echo were compared with sonograms of the quetzal—a magnificent bird venerated by the Maya and other Mesoamericans. The match was near-perfect! Was this a bizarre coincidence or a stunning archaeoacoustic discovery overlooked by archaeologists? The mathematics of the physical phenomena and the psychophysics of the chirped echo are now understood. Cultural evidence for intentional acoustical design by ancient Mesoamericans is strongly indicated. It seems certain that chirped echoes were the rule at staircased pyramids throughout Mesoamerica. Chirps weaken and become inaudible as limestone staircases erode and become porous.

10:15–10:30 Break

10:30

5aMU6. Traditional and recent performance practice in Asian free reed mouth organs: The sheng and khaen as case studies. James P. Cottingham (Phys., Coe College, 1220 First Ave., Cedar Rapids, IA 52402, jcotting@coe.edu)

Mouth-blown instruments employing a free reed coupled to a pipe resonator have long been used throughout East and Southeast Asia. Details of the origin and development of these instruments are not known, but are closely connected with the history and prehistory of a multitude of ethnic groups. Beginning from presumed folk instrument origins, the free reed mouth organs have been used in a variety of contexts including simple signaling, courtship, local entertainment, civic or military processions, and sophisticated court music. Two instruments operating on similar acoustical principles have had contrasting histories: the Chinese sheng and the Laotian khaen. The sheng has a two thousand year recorded history in China, and in the last century modernized versions have been developed and appeared in the Western concert hall style setting of the Chinese orchestra. The khaen, while remaining a strong cultural symbol of the Lao people, has not undergone similar developments as once prevalent traditional performance styles have almost disappeared.

10:50

5aMU7. Outdoor oratory and performance space. Braxton B. Boren (Mech. and Aerosp. Eng., Princeton Univ., 30-91 Crescent St., 5B, Astoria, New York 11102, bbb259@nyu.edu)

Spoken sermons and oratory in the West have usually been confined to the interior of churches or parliaments that bear similarities to traditional concert halls. However, the largest crowds in history are described as being outside due to the size constraints of most interior spaces. Historical estimates of crowd sizes or the intelligible range of speakers have generally been quite speculative. However, recent work has used an acoustical experiment by Benjamin Franklin to estimate the 1 m on-axis average SPL of the Anglican preacher

George Whitefield at 90 dBA. Computational simulation of Whitefield's preaching in London confirms Franklin's estimate that he could have been heard by crowds of 30,000 or more depending on weather conditions and crowd noise. Using Whitefield as a benchmark, other historical orations may be evaluated based on geometric and acoustic factors. Within this framework, speeches given by Pericles, Demosthenes, Alexander the Great, and Julius Caesar are examined based on the size and noise of the crowd as well as the level attainable by each speaker. In each case, outdoor oratory possesses unique performance considerations distinct from interior or musical settings.

11:10

5aMU8. The Paul's Cross sermons—Out of the cathedral and into the public square. Matthew Azevedo (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, mazevedo@acentech.com)

Paul's Cross, located in the northeastern yard of St. Paul's Cathedral, was the site of public sermons which were integral to the social fabric of early modern London. These sermons brought together a wide swath of London's population, "a great Congregation of thy Children... of all sorts... from the Lieutenant of thy Lieutenant, to the meanest sonne of thy sonne" in John Donne's words, for an event which served as a conduit from the King, through the Church, and out to the people of England. Early modern sermons were not merely lectures, but were theatrical conversations between the preacher and his congregation in which each had an active role to play, staged for their entertainment value as well as to elevate the spirit and the mind. The Paul's Cross Project attempts to understand this interplay between preacher and congregation in an open, public space through acoustical and visual modeling of St. Paul's churchyard as it existed before the Great Fire of 1666. The acoustical modeling has resulted in an auralization of the sermon which allows listeners to directly experience what it might have been like to attend one of the Paul's Cross sermons.

11:30

5aMU9. Recording studios—Optimized, contrived, and augmented spaces. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Multitrack production occurs around the globe in spaces from large to small. Their acoustic capability ranges from thoughtfully designed purpose-built spaces for recording to unmodified, re-purposed residential rooms, with everything in between. Yet recordings built entirely within a laptop-and-headphones production space have no problem sitting in shuffled playlists next to work from high end, world-class studios. Acoustic features—or lack thereof—are variously leveraged and overcome through recording craft and signal processing to adapt almost any production space to the sonic needs of the recording artist.

Session 5aNS

Noise and ASA Committee on Standards: Louis C. Sutherland's Lifetime Contributions to the Fields of Noise, Standardization, and Classroom Acoustics

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 101789, Germany

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

Chair's Introduction—8:30

Invited Papers

8:35

5aNS1. My collaboration with Louis C. Sutherland on classroom acoustic issues. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514, dlubman@dlacoustics.com)

8:55

5aNS2. Lou Sutherland—Bridging acoustics and people in acoustics. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

Lou Sutherland's scientific contributions have helped to renew the acoustic world. In all the years of my participation at ASA meetings, Lou has provided valuable insights that have advanced many subfields of noise. These include aircraft noise reduction, atmospheric sound propagation, classroom learning, and soundscape. Lou is always open to new ideas to improve acoustical calculations and measurements. His extensive work in classroom acoustics with David Lubman has increased his prominence in noise and architectural acoustics. For example, classroom acoustics has become one of the most persistent topics on the agendas of the Technical Committees on Noise and Architectural Acoustics for at least a decade. Although Lou and I have never worked together his publications on aircraft noise and soundscape have strongly influenced my research. Lou has an endearing and effective way of "inducting" new people into the ASA world. I remember with gratitude how Lou stimulated me to become active in ASA's Noise and Architecture Acoustics communities by introducing me to its distinguished members. I am very pleased for this opportunity to thank Lou for his past work. And to continue our valuable discussions that make the world a better place by reducing noise and improving soundscape.

9:15

5aNS3. Lou Sutherland—Contributions in aircraft noise. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooksacoustics.com)

Lou Sutherland has made pivotal contributions to the understanding of aircraft noise and its effects and to developing solutions for those problems. His work in noise effects ranges from the 707 aircraft prototype, the development of an engine noise suppressor for the B-52 to reduce sonic structural fatigue, to the Saturn rocket program. Lou's interests have encompassed aircraft engine turbomachinery cascade acoustic resonances and the propagation of sound through the atmosphere. These contributions were made while Lou was a researcher first at Boeing and later at Wyle Labs, as Chief Scientist and Deputy Director of Research. His insights have led to creative solutions to reduce aircraft noise effects on structures and on humans. He served as President of INCE and as an Associate Editor of JASA. In 2002 Lou was awarded the Acoustical Society of America Silver Medal in Noise "For contributions to the solution of aerospace and community noise problems, and for studies of molecular absorption and classroom acoustics." Most importantly, Lou has a terrific, infectious enthusiasm for all that he does, which is combined with a warm, generous spirit. Lou has provided kind and gentle leadership and guidance to many grateful beneficiaries over the years.

9:35

5aNS4. Lou Sutherland: A friend and colleague over the years. Paul D. Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

I have known Lou for a short time compared to some of you, only 50 years. Over these years, our paths have crossed in many ways at many times: reviewing the US Army's research program that I set up in 1972, development of structural response of buildings and its application to impulse noise and aircraft noise, joining with me and others in a collaborative consultative effort near the St. Louis airport, interacting with me as associate editor of JASA, and participating in Standards—notably the classroom acoustics standard.

9:55–10:15 Break

10:15

5aNS5. Lou Sutherland—The man and his work. Ben H. Sharp (7802 Trammell Rd., Annandale, VA 22003, bhs940@yahoo.com)

Lou Sutherland is one of the longest serving members of the international acoustics community and one of the most prolific. In a career lasting over 60 years, his main areas of research have involved studies of human and structural response to noise, sonic boom and blast, but his contributions to the fields of acoustics and vibration have been much more diverse and have included topics such as atmospheric absorption, aircraft noise measurement and modeling, low-frequency noise, and acoustic testing. He has been active in standards organizations and as a result of his services has been elected as a Fellow in several professional societies. As a consultant, Lou had the ability to break down complex problems into simple components that could be understood and solved, often using limited available data and simple theoretical models. Lou is still active as a consultant and continues to influence people with his ideas. As a long-term colleague, the author will examine some of his major contributions and share some of his experiences working with Lou.

10:35

5aNS6. Tribute to Lou Sutherland. Robert Kull (Marstel-Day, LLC, 203 Norton St., San Antonio, TX 78260, rkull@marstel-day.com)

This will be a brief tribute to Lou Sutherland who I first met in 1989. Although Lou Sutherland recognized that I had no acoustics background back then, he treated me with utmost respect and professionalism. Lou mentored, guided, and challenged me while I was program director for the Noise and Sonic Boom Impact Technology Advanced Development Program Office for the Air Force's Armstrong Medical Research Laboratory. I sought his wisdom as I faced critical decisions for my Service's research program. Over these past 25 years, we haven't had many occasions to work together, but the times we have had were significant and I will always cherish them.

10:55

5aNS7. Understanding speech in noisy rooms: Continuing the Sutherland legacy. Peggy B. Nelson (Univ. of Minnesota, 164 Pillsbury Dr. Se, Minneapolis, MN 55455, peggynelson@umn.edu)

Lou Sutherland has inspired us to continue to study the problems listeners experience when listening to signals in noisy rooms. We continue to follow his lead to learn more about the variability that adult and child listeners experience when listening in noise. Our current work allows adult listeners with hearing loss to self-adjust hearing aids in noisy rooms. Results suggest that hearing aid users vary significantly in the gain that they self-select for listening in noise. In general, as noise increases, most, but not all, listeners select less hearing aid gain. Overall, individual listeners varied greatly in their gain preferences even when hearing losses were very similar. These results suggest that we have much to learn about the processes of listening in complex environments. Variables will be discussed. The implications of this work suggest that a great deal is left unknown about performance in background noise. [Work supported by NIDCD R01 DC013267.]

11:15

5aNS8. Absorption of sound at high altitude: Lou Sutherland's contribution. Richard Raspet (NCPA, Univ. of MS, National Ctr. for Physical Acoust., University of MS, University, MS 38677, raspet@olemiss.edu), Andi Petculescu (Phys., Univ. of Louisiana Lafayette, Lafayette, MS), and Oleg A. Godin (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado, Boulder, CO)

Lou Sutherland was one of the original developers of ANSI S1-26, "Method for the calculation of the absorption of sound in the atmosphere." Research in the propagation of infrasound in support of the Comprehensive Test Ban Treaty Organization demonstrated the need for a reformulation of the calculation in the stratosphere and thermosphere. Lou and Hank Bass developed the formulation appropriate to the composition and environment of the upper atmosphere. The paper [J. Acoust. Soc. Am. **115**(3), 1012–1032 (2004)] documenting this work is an impressive example of the compilation of databases and ideas into a coherent model to predict the absorption of sound. In conclusion, we will briefly review recent research extending the analysis of infrasound in the upper atmosphere.

Session 5aSC

Speech Communication: Speech Perception and Production in Noise and Related to Disorders of Speech, Language or Hearing (Poster Session)

Noah H. Silbert, Chair
Communication Sciences & Disorders, University of Cincinnati, 3202 Eden Avenue, 344 French East Building, Cincinnati, OH 45267

Posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 8:15 a.m. to 9:45 a.m., and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m. There will be a 15-minute break from 9:45 a.m. to 10:00 a.m.

Contributed Papers

5aSC1. Affective prosody production in autistic and typically developed adult males. Daniel J. Hubbard, Daniel J. Faso, Noah J. Sasson, and Peter F. Assmann (School of Behavioral and Brain Sci., GR4.1, Univ. of Texas at Dallas, P.O. Box 830688, Richardson, TX 75083, dhubbard@utdallas.edu)

This study examined differences in production of affective prosody in adult males with autism spectrum disorder (ASD) and typically developed (TD) controls. Previous studies of children with ASD have reported increased variability in fundamental frequency (f0) in spontaneous and semi-spontaneous speech compared to TD children. A controlled set of expressive speech recordings was collected from 30 talkers (15 ASD) to measure differences between the two groups using the same lexical content. Isolated vowels, vowel-consonant-vowel (VCV) syllables, words and short phrases were elicited in five emotion contexts: angry, happy, interested, sad, and neutral. The recordings were obtained using evoked and portrayed elicitation techniques: talkers were asked to recall past emotional episodes (evoked) and role-play scripted scenarios (portrayed) specific to each emotion context. Consistent with previous work and extending the findings to adults producing the same lexical content, talkers with ASD showed increased f0 variability in each emotion context except for neutral. In addition, systematic group differences were found in acoustic properties other than f0 used to convey affective prosody—including harmonics-to-noise ratio and intensity—which were higher in each emotion context for talkers with ASD compared to TD talkers.

5aSC2. Production of contrastive focus in children with autistic spectrum disorder. lucile Rapin, Paméla Trudeau-Fisette, Marie Bellavance-Courtemanche, and Lucie Ménard (Linguist, UQAM, C.P. 8888, Succursale Centre-Ville, Montreal, Quebec H3C 3P8, Canada, lucilerapin@gmail.com)

Contrastive focus serves to emphasize the importance of a semantic unit in the language string. Children with autistic spectrum disorder (ASD) appear to show difficulties in producing this prosodic marker. This study aimed to identify acoustic correlates related to contrastive focus in children with ASD. Nine francophone children with ASD and nine francophone typically developing (TYP) children produced simple four-word sentences (for example, «C'est une chaise.»: «it is a chair.») in a neutral condition and then in a contrastive focus condition. Ninety-six speech productions were recorded using a system that synchronized acoustic signals with lingual and labial movements. Maximum pitch, mean pitch, and pitch range, as well as maximum and mean sound intensity and duration were investigated. Values for pitch range, maximum and mean sound intensity, and duration were greater in the focus condition than in the neutral condition. Moreover, the differences were significantly greater in TYP children than in ASD children, who did not have increased speech values when switching to the focus mode. This suggests that pitch range, and intensity and duration of sound correlate most with contrastive focus marking in both groups. Yet, it appears that ASD children show less contrastive focus marking than TYP children.

5aSC3. Fundamental frequency of speech directed to children who have hearing loss. Mark VanDam, Paul De Palma, and William E. Strong (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu)

Studies of child-directed speech (CDS) have shown that when talking to children, parents systematically use (among other strategies) increased fundamental frequency (F_0). Lombard effects such as increased F_0 have also been documented when addressing a listener who is hard-of-hearing (HH). Here, we examine F_0 of mothers and fathers in families with HH versus typically developing (TD) children in CDS and adult-directed speech (ADS) contexts. Whole-day audio recordings were collected by a child-worn audio recorder and analyzed by automatic speech recognition (ASR) software to identify segments of vocal activity by children and their parents (LENA Research Foundation, Boulder, CO). Custom software extracted F_0 values in all conditions. We found that (1) mothers are much more systematic in their use of CDS than fathers, (2) parents do not appear to be sensitive to the hearing status of their children, and (3) parents of HH children may have higher overall F_0 irrespective of CDS or ADS. Results suggest that mothers and fathers do not use F_0 with their children in the same way, and parents of children with hearing loss may have certain global F_0 characteristics not shared by parents of TD children.

5aSC4. Action verb processing correlates with motor asymmetry in Parkinson's disease. Emily Wang, Lee K. Walters (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 1611 West Harrison St., Ste. 530, Chicago, IL 60612, emily_wang@rush.edu), and Leo A. Verhagen Metman (Neurology, Rush Univ. Medical Ctr., Chicago, IL)

Onset of motor symptoms in PD is asymmetrical, and symptoms affect the side of onset more severely throughout the progression of the disease, which is known as "motor asymmetry." Previous studies revealed that PD patients with right-motor asymmetry (more severe symptoms on the right side of body) exhibited more speech impairment than those with left-motor asymmetry (Wang *et al.*, 2003; 2006). Patients with active deep brain stimulation (DBS) exhibited more severe speech deficits when they received either bilateral or left-only stimulation of the subthalamic nucleus (STN) than receiving right-only STN stimulation (Santens *et al.*, 2003). These findings lead to the current hypothesis that there is correlation between the motor asymmetry and linguistic processing of action verbs. Using a novel action verbal processing paradigm, 24 PD patients (12 right- and 12 left-motor asymmetry) and 11 age- gender-matched healthy controls were tested. The results showed that when the motor asymmetry was sufficiently different, i.e., when the difference of the UPDRS scores between the two sides of the body is greater than 2 (points), the right-motor asymmetry patients took

longer to process the action verbs, supporting the hypothesis that motor asymmetry is correlated with linguistic processing of action verbs at the cortical level.

5aSC5. Tongue movement pattern in speakers with dysarthria in production of "Ohio". Jimin Lee (Commun. Sci. and Disord., The Penn State Univ., 404A Ford Bldg., University Park, PA 16802, jxl91@psu.edu)

A different range of tongue motion during speech production has been assumed in speakers with dysarthria across different severity groups based on the previous findings in acoustic vowel space. To further confirm these findings, this study examined the tongue movement pattern in speakers with dysarthria during the production of "Ohio," a target word that requires the consecutive production of multiple vowels. Ten speakers with dysarthria of varying severity participated in this study. Three dimensional electromagnetic articulography (Wave system) was utilized to identify tongue movement trajectory during the target word production. Each speaker produced three repetitions of the word "Ohio" in the carrier phrase "I say a__ again." X, y, and z coordinate values from the tongue sensor (positioned approximately 25 mm from the tongue apex) were recorded. Statistical analyses showed that speakers with severe dysarthria produced a) less overall tongue movement, b) slower speed, c) less absolute tongue advancement-retraction and tongue height movement, and d) greater distance between starting /o/ and ending /o/ positions than speakers with mild dysarthria. Results are discussed with regard to the interpretation of tongue movement displacement and trajectory shape difference in speakers with dysarthria across different severity groups.

5aSC6. A comparative study of variability in landmark sequences and implications for dysphonic speech analysis. Keiko Ishikawa (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, 5371 Farmridge Way, Mason, OH 45040, ishikak@mail.uc.edu), Marepalli B. Rao (Dept. of Environ. Health, Univ. of Cincinnati, College of Medicine, Cincinnati, OH), and Suzanne Boyce (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Dysphonia negatively affects speech intelligibility especially in the presence of background noise. This may be because dysphonic speech often contains a higher proportion of noise and/or a lower proportion of harmonic power, leading to reduced information in the speech signal. Landmark (LM) analysis was designed to identify patterns of information in the speech signal that are particularly salient for the auditory system. Consequently, it describes speech as a sequence of LMs. Past studies successfully differentiated disordered speech from normal speech based on the number of times each LM occurs. While the count was a sufficient measure for their purposes, transitional patterns in LM sequences may yield more descriptive information on underlying mechanism of the intelligibility deficits. Shannon's Entropy and Markov chain model were used to evaluate the difference in LM sequences between normal and dysphonic speech. Landmarks were obtained from the first sentence of the Rainbow Passage for 33 normal speakers and 36 dysphonic speakers using SpeechMark^{TM} software package. The variability in the transitional patterns of LM was significantly less in dysphonic speech compared to normal speech. This suggests intelligibility deficits may be due to the greater acoustical constraints inherent to dysphonic speech.

5aSC7. The effect of Parkinson disease on voice onset time: Temporal differences in voicing contrast. Jason A. Whitfield (Speech Pathol. & Audiol., Kent State Univ., A104 Kent Ctr. for Performing Arts, Kent, OH 44242, jwhitfi4@kent.edu) and Alexander M. Goberman (Commun. Sci. and Disord., Bowling Green State Univ., Bowling Green, OH)

Parkinson disease (PD) affects the basal ganglia, which is involved with the selection, sequencing, and implementation of movement. Investigations suggest an extension of deficits to the speech motor and linguistic systems. Previous studies examining voice onset time (VOT) suggest that VOTs are neither systematically delayed nor systematically advanced in PD as compared to controls. The VOTs for voiced and voiceless stops are expected to differ for both individuals with PD and controls. However, inefficiencies in the sequencing and implementation of the voicing gesture may result in a smaller difference between VOT values of voiced and voiceless cognates. The current study examined VOT in individuals with PD and controls. Participants produced a corpus of stimuli to evaluate VOT using the carrier phrase "CVp again." The four corner vowels and all stop consonants were used. Speech was recorded and VOT values were manually measured for every utterance. Overall VOTs were significantly shorter for voiceless stops across all speakers. However, the average difference between the voice and voiceless cognates produced by the PD group was significantly smaller than the control group. These data suggest that the PD group exhibited less temporal distinction between voiced and voiceless stops.

5aSC8. Syllable position effects on perceptual determination of /r/ errors in expert and naive listeners. Sarah M. Hamilton, Keiko Ishikawa, Hedieh Hashemi Hosseinabad, Suzanne Boyce, and Lindsay Mullins (Commun. Sci. and Disord., Univ. of Cincinnati, 3433 Clifton Ave., Cincinnati, OH 45220, hamilsm@mail.uc.edu)

When ordinary listeners hear words containing a phoneme whose production differs in one feature from the target, the anomaly is often ignored if the word is recognizable. Phonetic research and clinical practice both depend on phonetics training to mitigate these top-down listening effects and enable detection of anomalies. Children with speech sound errors for American English /r/ are believed to have better productions when /r/ occurs at the beginning of a word. However, the shorter duration of /r/ word-initially may be leading clinicians to under-identify errors. While previous studies have shown that clinicians are better than naive listeners at identifying error /r/ in single syllables, they have not assessed identification of /r/ errors in whole words, where position may interact with lexical bias effects. In this study, speech-language pathologists and naive listeners rated children's natural speech productions of word-initial and word-final/r/ in whole words. These /r/ productions in words were selected to make a continuum defined by normalized third formant values. Reaction times and responses to stimuli were compared between listener groups. Initial results indicate that clinicians and naive listeners differ in their responses to formant profiles of stimuli, but that syllable position exerts a strong effect for both groups.

5aSC9. The use of visual information in non-native speech sound discrimination across the first year of life. D. Kyle Danielson, Padmapriya A. Kandhadai, and Janet F. Werker (Psych., Univ. of Br. Columbia, 2136 West Mall, Vancouver, British Columbia V6T 1Z4, Canada, kdanielson@psych.ubc.ca)

Infants are able to match seen and heard speech even in non-native languages, and familiarization to audiovisual speech appears to affect subsequent auditory-only discrimination of non-native speech sounds (Danielson *et al.*, 2013; 2014). However, the robustness of these behaviors appears to change rapidly within the first year of life. In this current set of studies, conducted with six-, nine-, and 10-month-old English-learning infants, we examine the developmental trajectory of audiovisual speech perception of non-native speech sounds. In the first place, we show that the tendency to detect a mismatch between heard and seen speech sounds in a non-native language changes across this short period in development, in tandem with the trajectory of auditory perceptual narrowing (Werker & Tees, 1984; Kuhl *et al.*, 1992; *inter alia*). Furthermore, we demonstrate that infants' familiarization to matching and mismatching audiovisual speech affects their auditory speech perception differently at various ages. While six-month-old infants' auditory speech perception appears to be malleable in the face of prior audiovisual familiarization, this tendency declines with age. The current set of studies is one of the first to utilize traditional looking-time measurements while also employing pupillometry as a correlate of infants' acoustic change detection (Hochmann & Papeo, 2014).

5aSC10. Effects of reading ability on lexically-informed perceptual learning. Emily Thompson, Stephen Graham, Alexandra T. Bohner, Julia R. Drouin, and Rachel M. Theodore (Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269-1085, emily.thompson@uconn.edu)

Research on perceptual learning for speech shows that lexical information can be used to modify phonological representations. Moreover, lexically informed perceptual learning is domain general in that it also influences the mapping to phonology from printed text. Given phonological processing deficits in adults with reading impairment, we hypothesized that reading ability would meditate how lexical information is used to dynamically adjust perceptual representations. Adult participants were identified as either good or poor readers based on standardized assessments of reading and reading sub-skills. All completed a lexical decision training phase followed by a test phase. Stimuli in both phases consisted of printed text. During training, readers viewed an ambiguous grapheme midway between "N" and "H." Lexical information was used to bias perception of the grapheme as either "N" or "H." At test, all readers categorized tokens along an "N" to "H" continuum. The results to date indicate that both groups of readers used lexical information to modify letter perception; however, the learning effect was more robust in the good readers compared to the poor readers. These results suggest that the degree to which lexical information is used to modify the mapping to sound structure is related to phonological ability.

5aSC11. Effect of intelligibility and speech rate on perceived listener effort. Kathleen Nagle (Speech & Hearing Sci., MGH Dept, of Laryngeal Surgery, Univ. of Washington, One Bowdoin Square, 11th Fl., Boston, MA 02114, kfnagle@uw.edu)

Perceived listener effort (PLE) has been defined as the perceived "amount of mental exertion required to attend to, and understand, an auditory system" (McGarrigle *et al*., 2014). PLE may be affected by multiple characteristics of the speech signal, including sentence length and intelligibility. Previous qualitative data indicates that listeners are also sensitive to speaking rate. This experiment measured the effect of speech rate on PLE for speech produced with a range of intelligibility. Eleven listeners with normal hearing (HINT), working memory (WAIS-IV), and receptive vocabulary (PPVT) transcribed and rated their PLE for the 144 low-context sentences produced with a monotone electrolarynx (EL) set at 75 Hz. Speech rate (syllables per second) was measured from onset of the first word to offset of the last word. Multiple regression analysis indicated that intelligibility and speech rate accounted for a significant amount of variance in PLE scores, $R^2 = 0.70$, $F(2,142) = 168.803$, $p < 0.001$, $R^2_{adj} = 0.70$. Intelligibility had a unique negative effect on PLE ratings ($b = -6.45$, $SE = 0.35$), $t(142) = -18.37$, $p < 0.001$, $sr^2 = 0.70$; however, rate was not uniquely predictive ($p > 0.05$). Future research should examine the association between radiated noise and PLE for EL speech.

5aSC12. Elevated depressive symptoms associate with an emotion-general deficit in speech perception at a cocktail party. Zilong Xie (Dept. of Commun. Sci. & Disord., The Univ. of Texas at Austin, 2504A Whitis Ave. (A1100), Austin, TX 78712, xzilong@gmail.com), W. Todd Maddox (Dept. of Psych., The Univ. of Texas at Austin, Austin, TX), and Bharath Chandrasekaran (Dept. of Commun. Sci. & Disord., The Univ. of Texas at Austin, Austin, TX)

Individuals with major depressive disorder demonstrate significant deficits in social communication. While the majority of work on individuals with elevated depressive symptoms focuses on speech production, much less is known about speech perception in individuals with elevated depressive symptoms. Here, we examine speech perception in the presence of different types of distractions. Two forms of distraction are hypothesized to interfere with speech perception in such listening conditions: energetic masking (EM) and information masking (IM). Relative to EM, IM places greater demands on executive function. A recent study showed that individuals with elevated depressive symptoms exhibit a selective deficit in perception of neutral speech during primarily IM conditions, likely due to their impairments in executive function. Here, we examined whether this selective deficit extends to emotional speech that portrays anger, fear, happiness, or sadness. Results showed that during IM conditions, individuals with elevated depressive symptoms exhibited poorer performance for neutral as well as for the four emotion categories, compared with those without elevated depression symptoms. However, both groups performed comparably during EM conditions. These findings suggest that elevated depressive symptoms are associated with an emotion-general deficit in speech perception in noise conditions that place demands on executive function.

5aSC13. An investigation into how the acoustics of different sized open plan classrooms affects speech perception in Kindergarten children. Kiri T. Mealings, Katherine Demuth (Linguist, Macquarie Univ., Level 3 AHH Macquarie Univ., Sydney, New South Wales 2936, Australia, kiri.mealings@students.mq.edu.au), Jörg Buchholz, and Harvey Dillon (National Acoust. Labs., Sydney, New South Wales, Australia)

Open plan classrooms, where several classes share the same space, have recently re-emerged in Australian primary schools. This paper examined how the acoustics of four Kindergarten classrooms (an enclosed classroom (25 students), a double classroom (44 students), a linear fully open plan triple classroom (91 students), and a semi-open plan K-6 classroom (205 students)) affect speech perception. Twenty-two to 23 children in each classroom participated in an online four-picture choice speech perception task while adjacent classes engaged in quiet versus noisy activities. The noise levels recorded during the task were higher in the larger open plan classrooms compared to the smaller classrooms for both the quiet and noisy conditions. A linear mixed effects model revealed that children's performance accuracy decreased as noise level increased. Additionally, children's speech perception abilities decreased the further away they were seated from the loudspeaker, and this effect was stronger the higher the noise level. Children's response time was also slower in the noisiest compared to quietest classroom. These results suggest that open plan classrooms may not be appropriate learning environments for didactic-style teaching with young children due to their high intrusive noise levels which negatively impact speech perception.

5aSC14. Using the stability of vocal onsets to evaluate vocal effort in response to changing acoustical conditions. Mark L. Berardi (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, markberardi12@gmail.com), Eric J. Hunter (Dept. of Commun. Sci. and Disord., Michigan State Univ., East Lansing, MI), and Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Several acoustical measures have been used in the past to evaluate vocal effort. They are useful in evaluating occupational risk for teachers or other occupational voice users. Analysis of vocal onsets has been used to show vocal effort in spasmodic dysphonia. Acoustical measures are also used in clinical speech-language pathology as an inexpensive and noninvasive ways of evaluating pathology severity and tracking therapy progress. In this presentation, a single acoustic parameter based on relative fundamental frequencies of glottal pulses following voiceless consonants (the onset coefficient) is used to evaluate vocal effort in response to changes in background noise and reverberation time within speaking environments. Analysis shows that males and females have similar vocal effort levels in the most typical acoustical conditions. However, females respond to louder background noise and longer reverberation times with more vocal effort than males.

5aSC15. Vocal comfort and effort in speech: Accommodation to different room acoustic conditions. Simone Graetzer, Eric J. Hunter, and Pasquale Bottalico (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, sgraetz@msu.edu)

Background: Vocal effort is a physiological entity that accounts for variation in voice production as loading increases, measured as sound pressure level (SPL). A number of studies have investigated vocal effort and load, but few have considered the role of acoustic clarity (C50), the early to late arriving sound ratio. Method: 20 subjects performed vocal tasks in various room acoustic conditions. Tasks were performed in three styles: soft style, comfortable conversational style, and loud classroom style. C50 in the position of the talker was changed by means of two reflective panels. Two noise levels were used: background noise, and artificial child babble noise. After each task, the subject answered questions addressing their perception of vocal comfort, control and fatigue. SPL and Fundamental Frequency (F0) were measured. Results: When panels were present, talkers perceived the room as being more comfortable to speak in. In particular, in the babble condition and in loud speech, comfort and control tended to increase. Vocal effort (SPL) tended to decrease when panels were present. An assessment of F0 is also reported. The results indicate that even while keeping reverberation time constant, reflective surfaces may be optimized to increase voice comfort and reduce vocal effort.

5aSC16. Vocal fatigue over a workday: A schoolteacher case study. Michael K. Rollins, Mark L. Berardi (Dept. of Phys. and Astronomy, Brigham Young Univ., 1371 N 380 W, Provo, UT 84604, michael.rollins@byu.edu), Eric J. Hunter (Dept. of Communicative Sci. & Disord., Michigan State Univ., East Lansing, MI), and Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Due to the vocal demands in their workplace and other factors, schoolteachers appear to be especially susceptible to long-term voice problems. Because female schoolteachers appear to engender an even higher risk, this case study focused on speech measures of one female elementary schoolteacher who reported feeling hoarse. The teacher was recorded over the course of a typical school day. Certain speech traits in the teacher's speech performance during elicitation tasks were evaluated by comparing changes from the beginning to the end of the school day. A strong correlation between the mean, standard deviation, skew, and kurtosis of long term average spectra (LTAS) and vocal fatigue was found, while other measures were found to be less consistent. The methods and findings will be discussed, and areas for further research suggested.

5aSC17. Vowel and sibilant sound production in noise. Kevin J. Reilly (Audiol. & Speech Pathol., Univ. of Tennessee, 12633 Wagon Wheel Circle, Knoxville, TN 37934, kreilly3@uthsc.edu)

The present study investigated speech production in noise and whether speakers modulate the spectral contrasts of vowel and sibilant sounds depending on the frequency characteristics of the background noise signal. Twelve speakers participated in the present study. Speakers were presented with three-word sequences one word at a time on a computer monitor and, after a variable delay, produced the three words in the order they were presented. Word sequences were comprised of CVC words containing either or both of the vowels, /a/ and /ae/, and either or both of the sibilants, /s/ and /ʃ/ , in word-initial position. During the presentation and production of each sequence, speakers were exposed to one of four possible noise conditions: (1) silence; (2) vowel masking; (3) sibilant masking; and (4) speech-shaped noise. The sibilant masker contained energy at frequencies associated with a speaker's productions of /s/ and /ʃ/ and the vowel masker at frequencies associated with their productions of /a/ and /ae/. These masking signals were generated prior to the experiment based on practice productions of the stimulus words by each speaker. Spectral contrast distances are based on vowel formant frequencies and sibilant spectral moments (1–4). Noise condition effects on spectral contrast distances are being evaluated to determine whether speakers selectively alter their production of those sounds masked by background noise.

5aSC18. Perceived emotional valence in clear and conversational speech for older adults with hearing loss. Shae D. Morgan and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

When asked to speak as though talking to an individual with hearing loss, talkers modify their speech from their everyday conversational style to a "clear" speaking style. Audiologists have observed in practice that individuals with hearing loss sometimes complain that their frequent communication partners are shouting at them, while the communication partners insist they are just trying to speak more clearly. A previous study investigated how individuals with normal hearing perceived emotional valence in clear and conversational speech and found that clear speech sounded angry more often than conversational speech. The present study will explore whether a similar effect is found in older individuals with hearing loss. Perceived anger has been partially attributed to increased energy in the high frequencies, where age-related hearing loss is most severe. Furthermore, aging effects have been found for visual emotion perception. Older adult listeners with hearing loss will be presented with conversational and clear sentences from the Ferguson Clear Speech Database (Ferguson, 2004) and asked to assign an emotion category to each sentence (anger, sadness, happiness, fear, disgust, or neutral). The results will indicate whether elderly listeners with hearing loss judge clear speech as sounding angry more often than typical conversational speech.

5aSC19. Speaking rate effects on phonemic boundary perception in cochlear implant users. Brittany N. Jaekel, Rochelle S. Newman, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland College Park, LeFrak Hall, College Park, MD 20742, jaekel@umd.edu)

Rate of speech can affect the durations of certain phonemes (Crystal & House, 1982), and listeners must normalize for speech rate to properly identify words (Miller, 1981). How speech rate affects perception of phonemic boundaries in users of cochlear implants (CIs) in comparison to normal-hearing (NH) listeners is unknown. The speech processing that occurs in these devices may obscure word and phoneme boundaries, providing less reliable durational information about the incoming speech signal. Less effective rate normalization in CI users could thus contribute to the degraded speech perception often observed in this population. Preliminary results with stop-consonant series show absolute differences between NH and CI listeners in the precise location of phonemic boundaries (with CI users' boundaries typically occurring at shorter voice onset time durations), but not in the relative relationship of these boundaries as a function of speech rate. Data from NH listeners presented a CI simulation show effects of number of channels on both the location of phonemic boundaries and presence of rate normalization.

5aSC20. Effect of context on bimodal benefit for temporally interrupted sentences in simulated electric-acoustic stimulation. Soo Hee Oh, Gail Donaldson (Univ. of South Florida, Tampa, FL), and Ying-Yee Kong (Northeastern Univ., 226 Forsyth Bldg., 360 Huntington Ave., Boston, MA 02115, yykong@neu.edu)

Previous research has demonstrated a clear effect of linguistic context on bimodal benefit for continuous speech, implicating an interaction between low-frequency acoustic cues and top-down linguistic processing (Kong *et al.*, *CIAP 2013*). However, bimodal benefit appears to be reduced for phonemic restoration (Baskent, 2012 *JARO*), suggesting that low-frequency facilitation of top-down processing is weakened when the speech stream is interrupted. The present study examined top-down effects in temporally interrupted speech by comparing bimodal benefit for low- and high-context sentences. Young, normal-hearing listeners were presented with City University of New York (CUNY) or Institute of Electrical and Electronics and Engineers (IEEE) sentences that were gated with silence (5 Hz, 50% duty cycle). One ear received sentences that were noise-band vocoded (8, 12, or 16 channels for CUNY; 12, 16, or 32 channels for IEEE) and the other ear received low-pass (LP) speech or LP harmonic complexes (LPHCs). Findings demonstrated clear effects of context on bimodal benefit when LP speech was presented to the residual-hearing ear, however, the benefits observed were considerably smaller than those reported previously for continuous speech. Unlike previous findings for continuous speech, no bimodal benefits were observed when LPHCs were presented to the LP ear.

5aSC21. Cutaneous vibration enhances the Lombard Effect. François-Xavier Brajot (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover W221, Athens, OH 45701, brajot@ohio.edu) and Vincent L. Gracco (Commun. Sci. and Disord., McGill Univ., Montreal, Quebec, Canada)

The Lombard Effect is traditionally defined as a compensatory vocal response to noise at the ear. Somatosensory feedback is also relevant to speech, however, and may be expected to play a role in the Lombard response and perceived vocal effort. The effect of masking both auditory and laryngeal vibrotactile feedback was assessed in adults without speech or hearing disorders. Autophonic loudness, speech intensity, fundamental frequency, and formant frequencies were measured in each of three conditions: unmasked, auditory masking, and mixed auditory plus somatosensory masking. Neither the slope of the autophonic loudness curve nor the shape of the oral reading intensity contour changed as a function of masking condition.

However, the intercept of the autophonic loudness curve and the offset of the intensity contour were greater in the mixed masking as compared to the auditory masking condition, indicating that vibrotactile masking effectively enhanced the Lombard Effect. These findings support the hypothesis that both auditory and somatosensory feedback function to adjust speech gain, without affecting relative production intensity or autophonic loudness perception. As such, the Lombard Effect is proposed to result from a general sensorimotor process rather than from a specific audio-vocal mechanism.

5aSC22. Segmental interference by the masker modulation spectrum in sentences: Effects of age and hearing loss. Daniel Fogerty (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, University of South Carolina, Keenan Bldg., Ste. 300, Columbia, SC 29208, fogerty@sc.edu), Jayne B. Ahlstrom, William J. Bologna, and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Younger normal-hearing, older normal-hearing, and older hearing-impaired subjects listened to sentences with consonants or vowels replaced with noise modulated by the envelope of a competing talker. The modulation spectrum of the noise was low-pass filtered at different cutoff frequencies. Sentences were spectrally shaped according to listeners' individual audiometric thresholds to ensure sufficient audibility; in addition, a second group of younger normal-hearing subjects listened to sentences that were spectrally shaped according to the mean audiogram of the hearing-impaired subjects. Results demonstrated declines in sentence intelligibility for older as compared to younger listeners, with more evident declines for older hearing-impaired listeners. Declines in sentence intelligibility based on vowel cues with increasing modulation rates in the competing noise were noted for all listener groups. In contrast, sentence intelligibility based on consonant cues increased when the competing noise included modulation rates between 8 and 16 Hz. An adverse effect of spectral shaping was also observed, with an overall decrease in performance for younger spectrally matched listeners. Thus, in temporally complex listening conditions, spectral shaping to assure audibility may not fully control for reduced sentence intelligibility among hearing-impaired listeners and may interact with temporal properties of the speech signal. [Work supported by NIH/NIDCD and ASHA.]

5aSC23. Phonemic restoration with envelope and periodicity cues: Effects of age and competing talkers. William J. Bologna (Dept. of Hearing and Speech Sci., Univ. of Maryland, 135 Rutledge Ave., MSC 550, Charleston, SC 29425, bologna@musc.edu), Monita Chatterjee (Auditory Prostheses & Percept. Lab., Boys Town National Res. Hospital, Omaha, NE), and Judy R. Dubno (Dept. of Otolaryngol. - Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Perception of interrupted speech improves when noise is inserted in silent gaps between speech segments. This "phonemic restoration" is enhanced with intervening noise that contains low-level speech information, such as noise modulated by the temporal envelope of the missing speech. However, this benefit may be reduced in multiple-talker environments where competing speech contributes additional masking effects. To minimize these masking effects, periodicity information within intervening noise segments of interrupted speech may help listeners segregate multiple voices and thus increase phonemic restoration. The relative benefit of envelope and periodicity cues, and their use by older adults with poorer performance in multiple-talker environments, remains unclear. To address these questions, younger and older adults with normal hearing listened to target sentences in quiet and competing talker backgrounds; target sentences were periodically interrupted with silence, envelope-modulated noise, or pulse trains, which contained periodicity information from the missing speech. Phonemic restoration was defined as the difference in recognition of interrupted sentences filled with envelope-modulated noise or pulse trains as compared to sentences interrupted by silence. Results are discussed in terms of contributions of envelope and periodicity cues to perceptual organization in complex listening environments. [Work supported by NIH/NIDCD and a AAA Student Investigator Research Grant.]

5aSC24. Across-formant integration and speech intelligibility: Effects of acoustic source properties in the presence and absence of a contralateral interferer. Robert J. Summers, Brian Roberts (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, r.j.summers@aston.ac.uk), and Peter J. Bailey (Dept. of Psych., Univ. of York, York, United Kingdom)

Three experiments used three-formant (F1 + F2 + F3) analogues of natural sentences to explore the role of acoustic source properties in across-formant integration. In experiment 1, F1 + F3 were generated using a monotonous periodic source (F0 = 140 Hz) and second-order resonators (H1 + H3); in experiment 2, F1 + F3 were tonal analogues (T1 + T3). F2 could take either form (H2 or T2). Target formants were always presented monaurally; the target ear was assigned randomly on each trial. In some conditions, only the target was present; in others, a competitor for F2 (F2C) was added contralaterally. Listeners must reject F2C to optimize recognition. Competitors (H2C or T2C) were created using the time-reversed frequency and amplitude contours of F2. Without F2C, intelligibility was reasonably high and the effect of a source mismatch between F1 + F3 and F2 was negligible. For both shared- and hybrid-source targets, the impact of adding F2C was modest when it was tonal but large when it was harmonic, irrespective of whether F2C matched F1 + F3. Experiment 3 showed that this pattern was maintained when corresponding harmonic and tonal analogues were loudness-matched. These findings extend those from earlier research using dichotic targets. Source type and competition, rather than acoustic similarity, govern the phonetic contribution of a formant. [Work supported by ESRC.]

5aSC25. Informational masking of monaural speech by a single contralateral formant. Brian Roberts and Robert J. Summers (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk)

Recent research suggests that the ability of an extraneous formant to impair intelligibility depends on the variation of its frequency contour. This idea was explored using a method that ensures interference occurs only through informational masking. Three-formant analogues of sentences were synthesized using a monotonous periodic source (F0 = 140 Hz). Target formants were presented monaurally; the target ear was assigned randomly on each trial. A competitor for F2 (F2C) was presented contralaterally; listeners must reject F2C to optimize recognition. In experiment 1, F2Cs with various frequency and amplitude contours were used. F2Cs with time-varying frequency contours were effective competitors; constant-frequency F2Cs had far less impact. Amplitude contour also influenced competitor impact; this effect was additive. In experiment 2, F2Cs were created by inverting the F2 frequency contour about its geometric mean and varying its depth of variation over a range from constant to twice the original (0–200%). The impact on intelligibility was least for constant F2Cs and increased up to ~100% depth, but little thereafter. The effect of an extraneous formant depends primarily on its frequency contour; interference increases as the depth of variation is increased until the range exceeds that typical for F2 in natural speech. [Work supported by ESRC.]

5aSC26. Effect of noise on foreign-accent adaptation. Elisa Ferracane, Cynthia P. Blanco, Gabriela Cook, Karen Johnson, and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, 305 E. 23rd St., Austin, TX 78712, elisa@ferracane.com)

Previous work indicates that listeners are able to adapt quickly to changes in accent [Clarke & Garrett, 2004; Bradlow & Bent, 2008] and that adaptation is faster for familiar accents [Witteman *et al*., 2013; Blanco *et al*., 2014]. The present study examines how listeners adapt to more and less familiar accents presented in noise as compared to quiet, since noisy listening environments are more perceptually challenging [Vaden *et al*., 2013; Gilbert *et al*., 2014] but also more closely represent typical listening conditions. Native English listeners heard blocks of sentences produced by native- or foreign-accented talkers (Korean, Spanish) and responded to a visual probe. Listener accuracy and reaction times were compared across

accent and speaker blocks. Results showed that, unlike in quiet, listeners were unable to adapt to the less-familiar Korean accent. Processing of the familiar Spanish accent was successful in noise and quiet. Adaptation in noise was furthermore correlated with talker intelligibility: less intelligible talkers elicited lower accuracy and slower response; in quiet, there was no talker effect. The results revealed that the ability to adapt to a novel foreign accent is disrupted, as the extra perceptual effort in noise prevents listeners from generalizing about systematic variability of foreign-accented features.

5aSC27. Gap detection and speech recognition in noise in younger versus older listeners. Susan M. DeMetropolis, Janet R. Schoepflin, and Lawrence J. Raphael (Commun. Sci. and Disord., Adelphi Univ., 158 Cambridge Ave., Garden City, NY 11530, susandemetropolis@mail.adelphi.edu)

The ability to encode information in temporal envelopes of acoustic stimuli is an important skill of the auditory system. It has been demonstrated that a listener's temporal-processing capability is predictive of performance on speech recognition, especially in noisy and complex environments (George, Festen, & Houtgast, 2006; George *et al*., 2007; Shen, 2014; Snell, Mapes, Hickman, & Frisina, 2002). The goal of this study was to determine whether temporal acuity using gap detection thresholds from the Gap-in Noise (GIN) test (Musiek, 2003) has any relation to the perception of speech in noise from the Revised Speech Perception in Noise (R-SPIN; Bilger, 1984). All younger listeners were between 21 and 30 years (n = 7/mean = 24.7) and older listeners (n = 4/mean = 67.25) were between 65 and 71 years old with audiometric thresholds equal or better than 15 dB HL between 250 and 8000 Hz in both ears, monolingual English, and intact cognitive skills. Results revealed that temporal acuity of older listeners was reduced compared to the younger listeners. Despite temporal processing differences between the two age groups on the GIN test, they performed similarly on the R-SPIN. Results suggest that impaired abilities to process envelope and fine structure cues of acoustic signals does not influence speech perception in background noise.

FRIDAY MORNING, 22 MAY 2015 — KINGS 1, 8:30 A.M. TO 11:00 A.M.

Session 5aSP

Signal Processing in Acoustics: Detection, Classification, and Analysis

Brian E. Anderson, Cochair
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Contributed Papers

8:30

5aSP1. Performance analysis of a fuzzy soft decision CFAR detector in non-Rayleigh background. Yanwei Xu, Chaohuan Hou, Shefeng Yan, and Xiaochuan Ma (IOA CAS, Beisihuanxilu NO.21, Beijing 100191, China, xyw@mail.ioa.ac.cn)

A fuzzy statistical normalization fuzzy constant false alarm rate (FSN-FCFAR) detector in non-Rayleigh background based on fuzzy statistical normalization and fuzzy soft decision is proposed. The performance of the proposed fuzzy soft decision detector is studied both for homogeneous backgrounds and for non-homogeneous environments caused by interfering targets or clutter edges. Performance comparisons with the conventional hard decision CFAR detectors such as CA-CFAR, GO-CFAR and OS-CFAR are carried out. The comparison results show that the proposed FSN-FCFAR detector can not only get a very good detection performance in homogeneous backgrounds, but also can confront interfering targets and clutter edges at the same time in non-homogeneous environments. Moreover, the fuzzy soft decision detector can provide more valuable information than the hard decision detector for data fusion, target tracking or object identification.

8:45

5aSP2. Lucky ranging in underwater acoustic environment subject to spatial coherence loss. Hongya Ge (Elec. & Comput. Eng., New Jersey Inst. of Technol., University Heights, Newark, NJ 07102, ge@njit.edu) and Ivars P. Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

In principle, a passive sonar system utilizing a long line array can estimate a source's range by measuring its wavefront curvature. However, implementation can be difficult because wavefront curvature ranging is highly sensitive to spatial coherence losses. Real-world data measured by towed arrays often exhibit rapid fluctuations in signal spatial coherence across time and frequency. Such non-stationary degradation in spatial coherence on distributed arrays bears the similarity to the atmospheric Kolmogorov Turbulence Effect present in ground based telescopes (blurring of stars). Inspired by the lucky imaging approach used in astronomy, we propose a new approach for passive ranging in underwater environments subject to time-varying spatial coherence loss. Over short time scales, we speculate that the ocean may be well-behaved in the sense that there are little or no distortion to the signal wavefronts aka lucky moments. If detected, these lucky moments can be utilized to estimate range more accurately, overcoming coherence loss degradations experienced over long integration times. Modeling the signal as either coherent or incoherent in a short time frame, we derive the maximum likelihood lucky range estimator and show that with proper time-scale and processing bands, lucky moments can be identified that yield greatly improved estimates of range.

9:00

5aSP3. Classification of signals in spherically invariant random clutter. Bruce K. Newhall and Anna Slowikowski (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723, bruce.newhall@jhuapl.edu)

The generalized likelihood ratio test (GLRT) is derived for the case of a signal subspace in acoustic clutter characterized by a spherically invariant random variable (SIRV). This result is a generalization of two previous results. First, the GLRT for the detection of a one dimensional signal in

SIRV clutter has been previously given. However, featureless classification work has previously considered signals that are members of a multidimensional subspace but only in Gaussian clutter. The SIRV model extends that to the non-Gaussian clutter case. A SIRV is the product of two random variables: a non-Gaussian scalar times a complex multivariate Gaussian vector. The general GLRT result is then applied to a generalized gamma distribution for the SIRV scalar. When the generalized gamma exponent parameter is -2 this produces a product SIRV whose acoustic intensity has a generalized Pareto distribution. When the exponent parameter is $+2$, the SIRV intensity is k distributed. This demonstrates that two widely used clutter distribution models are special cases of this more general distribution. Methods of parameter estimation for application of the technique are also given. [Work supported by the Office of Naval Research.]

9:15

5aSP4. An approximation method for dispersive wave propagation. Jonathan Ben-Benjamin, Leon Cohen (Dept. of Phys., Hunter College, 695 Park Ave., New York, NY, yonatan@greatwing.com), and Patrick Loughlin (BioEng., Univ. of Pittsburgh, Pittsburgh, PA)

A phase space approximation method that extends a previous single-mode approximation [Loughlin, Cohen, J. Acoust. Soc. Am. **118**, 1268 (2005)] for linear dispersive wave propagation is developed. We show that each mode is governed by a Schrodinger-type equation, where the corresponding Hamiltonian operator is non-Hermitian if the dispersion relation is complex, for which case there is absorption. The propagated wave is obtained by evolving each mode according to its respective Schrodinger equation. We show how to obtain the initial modes from the initial conditions on the wave. We then formulate the propagation problem in phase space and obtain the exact equation of motion for the phase space function. We also obtain an approximate solution for the phase space evolution of the wave, which involves a simple substitution into the initial phase space function. Examples are given for a parallel plate wave guide and the beam equation. [Work supported by ONR, code 321US.]

9:30

5aSP5. Environmentally corrected matched filter. H. John Camin (Graduate Program in Acoust., Penn State Univ., ARL Penn State Univ., P.O. Box 30, M.S. 9310L, State College, PA 16804, jcamin@psu.edu)

Matched filtering is commonly used to process broadband active acoustic echoes providing higher levels of range resolution and reverberation noise suppression than can be realized through narrowband processing. Since theoretical processing gains are proportional to the signal bandwidth, it is typically desirable to utilize the widest band signals possible. However, as signal bandwidth increases, so do environmental effects that increase decorrelation between the received echo and the transmitted waveform. This loss of coherence often results in processing gains and range resolution much lower than theoretically predicted. Weiner filtering, commonly used in image processing to improve distorted and noisy photos, was investigated as an approach to correct for these environmental effects. This improved signal processing, environmentally corrected matched filter (ECMF), first uses a Wiener filter to estimate the environmental transfer function and then again to correct the received signal. This process can be viewed as a "smarter" inverse or whitening filter that optimally adjusts according to the signal to noise ratio across the spectrum. Sonar simulation toolset (SST) synthetic data and measured in-air data will be presented to illustrate the improved processing.

9:45

5aSP6. Three component vibrational time reversal communication. Brian E. Anderson, Timothy J. Ulrich, and James A. Ten Cate (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, bea@lanl.gov)

Time reversal provides an optimal prefilter matched signal to apply to a communication signal before signal transmission. Time reversal allows compensation for wave speed dispersion and can function well in reverberant environments. Time reversal can be used to focus elastic energy to each of the three components of motion independently. A pipe encased in concrete was used to demonstrate the ability to conduct communications of information using three component time reversal. The ability of time reversal to compensate for multi-path distortion (overcoming reverberation) will be demonstrated and the rate of signal communication will be presented. [The U.S. Department of Energy, through the LANL/LDRD Program, is gratefully acknowledged for supporting this work.]

10:00

5aSP7. Analysis of signal detection SNR limits in snapshot-deficient scenarios with colored noise. Jose A. Diaz-Santos (Naval Surface Warfare Ctr., 18444 Frontage Rd., Ste. 323, Dahlgren, VA 22448-5161, jose.diaz-santos@navy.mil) and Kathleen E. Wage (George Mason Univ., Fairfax, VA)

Most source enumeration algorithms assume a white noise background; however, in many underwater applications, the background noise is colored. Various researchers have investigated the problem of source enumeration when the background noise is colored. The standard approach is to apply a whitening filter before the signal enumeration algorithm. Most of these algorithms perform poorly when the number of noise snapshots used to estimate the whitening filter is small. Nadakuditi and Silverstein [IEEE J. Sel. Topics Signal Process., 2010] developed an algorithm for source enumeration using random matrix theory that provides the fundamental SNR limits for snapshot-deficient scenarios with arbitrary noise. Nadakuditi and Silverstein's analysis focuses on the performance when the number of signal plus noise snapshots varies while the number of snapshots used for the whitening filter stays constant. This talk analyzes the performance of this algorithm when number of snapshots available to estimate the whitening filter varies from N to 10 N. Simulations using two different colored noise models for a large vertical linear array located in the deep ocean will be presented.

10:15

5aSP8. Effect of reverberation on instrumental vibrato tones. Sarah R. Smith and Mark F. Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, 405 Comput. Studies Bldg., P.O. Box 270231, Rochester, NY 14627, sarahsmith@rochester.edu)

Musical vibrato is the process by which a performer subtly varies the frequency and amplitude of a note for expressive effect. Previous work has suggested that the vibrato modulation may not be uniform across all overtones, causing the individual frequency trajectories to become decorrelated. In this study, we investigate the effect of reverberation on instrumental tones performed with vibrato. This is accomplished by performing a detailed time-frequency analysis to extract the instantaneous frequencies of many overtones. In order to accurately track the low amplitudes of the upper harmonics, we present an algorithm tailored for optimal tracking in as little as 10 dB SNR. For tones recorded in an anechoic environment, the instantaneous frequency modulations of all the overtones are found to be highly correlated. However, tones recorded in a reverberant space, as well as anechoic recordings processed with a synthetic reverberation are found to have significantly lower correlations between the overtones. In addition to providing insight into musical timbre variations, these results could be useful for characterization of acoustic transfer functions or applications where overtone correlation is assumed, such as audio source separation.

10:30

5aSP9. Time delay estimation of acoustic tones with unpredictable frequency variations under low signal to noise ratio conditions. Stephen J. Franklin, Anthony Finn, and Joshua Meade (School of Eng., Univ. of South Australia, W2-56 Mawson Lakes Campus, UniSA, Mawson Lakes, South Australia 5095, Australia, stephen.franklin@unisa.edu.au)

This paper describes a novel approach for autonomously detecting and tracking aircraft in the vicinity of a unmanned aerial vehicle (UAV). The time difference of arrival (TDOA) of acoustic tones emanating from the distant aircraft are correlated between spatially distributed microphone pairs located on the UAV. The geometry of multiple microphone pairs then allows the elevation and azimuth of the approaching aircraft to be estimated, despite the high levels of onboard narrow- and broadband noise emanating from the engine firing sequence, propeller, airflow over the microphones, and mechanical vibration. Current signal processing techniques estimate time delay when the signal level of the approaching aircraft is above the background signature of the sensing aircraft, despite subtle frequency variations in the narrowband components of the approaching aircraft's signature.

The technique described in this paper is designed to track these unpredictable frequency variations and thus extends the detection range of the approach by enabling estimation of TDOA when the signal level is 20 dB below the noise floor. Potential detection ranges in excess of 1 km are demonstrated when the signal processing is combined with careful suppression of the many noise sources onboard the UAV.

10:45

5aSP10. A novel high-resolution algorithm for separating ray paths interrupted by colorful noise. Longyu Jiang (Southeast Univ., Sipailou 2#, Nanjing 210096, China, JLY01412@gmail.com)

Abstract: In the first step of ocean acoustic tomography, ray paths need to be identified with parameters which will be utilized in inversion process. Many second-order direction finding algorithms have been applied to identify ray paths in this context, such as beamforming, MUSIC algorithm. Because ray paths are produced by reflection and (or) refraction of emitted signal, they are fully correlated or coherent. Recently, the smoothing-MUSICAL algorithm, which is based on second-order statistics, has been developed to separate fully correlated or coherent signals and gained large improvement of separation ability in DOA-temporal domain. All the above methods work under the assumption of white Gaussian noise, however, in the real ocean environment there always exists colorful noise. In this paper, we propose a high-resolution algorithm based on four-order cumulants, which enables to separate fully correlated or coherent signals interrupted by colorful noise. Its performance has been illustrated by synthetic experiments and real experiments.

FRIDAY MORNING, 22 MAY 2015 — BALLROOM 4, 8:00 A.M. TO 11:30 A.M.

Session 5aUW

Underwater Acoustics: Propagation, Tomography, Scattering, and Transducers

Joseph D. Schneiderwind, Cochair
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Brandon Patterson, Cochair
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Contributed Papers

8:00

5aUW1. Striation processing of continuously active sonar data. Scott Schecklman and Lisa M. Zurk (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, sscheck@pdx.edu)

Previous work by the authors has shown that striations associated with the waveguide invariant can provide a new physics-based tracking constraint that improves tracking performance in pulsed active sonar data. A necessary preprocessing step is to isolate the target return from the direct arrival and other target echoes. Striation-based beamforming can be used as an optional preprocessing step to preserve the multipath structure in the beamformer output of a horizontal line array. Recent interest in continuously active sonar (CAS) processing has motivated research to determine how striations might be similarly exploited for tracking when the sonar is operated with a high duty cycle. In CAS signals, the direct arrival is almost always overlapping in time with the target echo return and the standard time-gating technique cannot isolate the echo return. In this paper, a time-dependent filter is introduced to isolate and extract the CAS echo spectrum. The conditions in which striation-based beamforming should be applied to preserve the striation structure in the beam output are also explored. The physics-based signal processing algorithms discussed here are expected to provide valuable information about the target's multipath structure, which can be exploited in an extended Kalman filter for enhanced tracker performance.

8:15

5aUW2. Comparisons between energy-flux propagation models utilizing single- and multiple-scattering treatments of sea-surface roughness in a range-independent waveguide. Joseph D. Schneiderwind, Derek Olson, Charles W. Holland, and Anthony Lyons (Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, jds563@psu.edu)

Acoustic scattering from a rough sea surface in propagation modeling is generally treated using a single-scatter approximation in which only specular components are considered and the incoherent field is neglected. This work presents energy-flux calculations of transmission loss for a shallow water waveguide with a rough sea-surface following a Pierson-Moskowitz spectrum. Multiple scattering is evoked by considering conservation of energy from scattering events. Comparisons are made between the model with multiple scattering and that using the single-scatter approximation. Models are compared to the direct solution using the boundary element method. [This work was partially funded by the Department of Defense (DoD) through the National Defense Science & Engineering Graduate Fellowship (NDSEG) Program, the Eric Walker Fellowship from the Applied Research Laboratory at the Pennsylvania State University, and the Achievement Rewards for College Scientists (ARCS) Foundation Pittsburgh Chapter.]

8:30

5aUW3. A follow up low frequency propagation experiment in Currituck Sound. Richard D. Costley (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, dan.costley@usace.army.mil), Andrew R. McNeese, Megan S. Ballard, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas, Austin, TX), Kent K. Hathaway (Coastal and Hydraulics Lab., US Army Engineer Res. & Development, Kitty Hawk, NC), Eric Smth (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., Vicksburg, MS), Preston S. Wilson, and Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas, Austin, TX)

An initial low frequency propagation experiment was conducted in September 2013 in which a Combustive Sound Source (CSS) generated transient acoustic signals in water depths of approximately 2.5 m [Costley *et al.* J. Acoust. Soc. Am. **136**, 2178 (2014)]. High amplitude signals were generated mid-water column and recorded with bottom mounted hydrophones at distances up to 200 m. A time harmonic, two-dimensional axisymmetric finite element model under-predicted the magnitudes of the recorded pressure signals. A follow up experiment was conducted in October 2014. In addition to using the CSS and the bottom mounted hydrophones, a 4-element vertical line array was deployed. Measurements were made at source-receiver separations from approximately 10 m to 1000 m in 100 m increments. Additionally, acoustic properties of the sediment were obtained through in-situ measurements [McNeese *et al*. J. Acoust. Soc. Am. **136**, 2252 (2014)]. The results of the experiment, along with results of the finite element model using updated sediment properties, will be presented. [Work supported by the U.S. Army Engineer Research and Development Center. Permission to publish was granted by Director, Geotechnical & Structures Laboratory.]

8:45

5aUW4. Efficient estimation of the probability density function of transmission loss in uncertain ocean environments. Brandon Patterson and David R. Dowling (Univ. of Michigan, 626 Spring St., Apt. #1, Ann Arbor, MI 48103-3200, awesome@umich.edu)

Predictions of acoustic transmission loss (TL) in the ocean are useful in a variety of naval and ocean engineering applications. However, real ocean environments are imperfectly known and have many uncertainties. Consequently, TL calculations performed for these environments are also uncertain. Traditional Monte-Carlo methods used to obtain probability density functions (PDF) of TL in such circumstances require many field calculations, and are impractical for real-time applications. This presentation describes how the PDF of TL at a point of interest may be estimated using a single TL-field calculation and the statistics of TL within a range-depth area containing that point. To document the performance of this technique, RAM was used to perform TL field calculations for 100 Hz and 300 Hz sound sources in realistic ocean sound channels with eight uncertain environmental parameters describing the bathymetry, sound speed profile, and bottom properties. Preliminary results indicate that TL PDFs generated with area statistics have an L_1-error $\leq$ 0.5 when compared with 1000-calculation Monte-Carlo PDF results throughout most of the sound channel. However, greater errors were observed in the near field as well as in areas where the sound channel depth was less than 100 m, for 100 Hz sound. [Sponsored by ONR.]

9:00

5aUW5. Scattering of plane acoustic waves at elastic particles with rough surfaces. Leif Bjorno (Stendiget 19, Taastrup 2630, Denmark, prof.lb@mail.dk)

A comprehensive theoretical and numerical study of the influence of surface roughness of elastic particles in water on the scattering of ultrasonic waves has been carried out. For near spherical shape of the particles and with small rms-roughness heights, a perturbation method has been developed. In this method, the first-order perturbation contribution predicts the contribution to the incoherently scattered acoustic field due to surface roughness, and the second-order perturbation contribution predicts the change in the coherent field and will satisfy the requirement of energy conservation. The second-order perturbation contribution is evaluated by use of the form function concept, while the first-order perturbation to the total scattered acoustic field is evaluated by use of the scattering cross-section. As a function of the ka-value and for different rms-roughness heights a numerical study of the forward and backward scattering from rough, elastic particles has been carried out and a substantial roughness influence on the scattered field has been verified. Some experimental results from measurements of scattering from glass and cast iron spheres have given evidence to the numerical predictions.

9:15

5aUW6. Measured backscattering of a first order vortex beam by a sphere with helicity selective processing and imaging. Viktor Bollen, Daniel S. Plotnick, David J. Zartman, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, viktor.bollen@wsu.edu)

An acoustic vortex beam's wavefield has a null on the axis of propagation and an angular phase ramp. A four-element transducer generates a first order vortex beam in water by appropriately phasing each element [V. Bollen *et al.*, Proc. Meet. Acoust. **19**, 070075 (2013)]. A differential beam having a broader null may also be produced by exciting two elements out-of-phase. The four-element transducer was also used to measure backscattering by a small solid sphere scanned in a raster pattern and insonified by the vortex beam. By independently recording each of the four elements and adding appropriate time delays in post-processing, the scattering may be measured in a helicity neutral, and co- and cross-helicity sensitive modes; this allows the retention of the vortex source without modifying the experimental setup. Due to the rapid phase change on the axis of the beam, subwavelength resolution of the null location of the sphere can be achieved. By applying a time delay-and-sum imaging algorithm this resolution can be further increased. Three-dimensional images can also be constructed from the data. By driving two opposing elements, the tilt of the transducer with respect to the scanning plane can be inferred, aiding alignment. [Work supported by ONR.]

9:30

5aUW7. Bubble nonlinear oscillation in sound field. Desen Yang, Shiyuan Jin, Shengguo Shi, Jie Shi, and Haoyang Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145 Nantong St., Nangang District, Heilongjiang, Harbin 150001, China, jinshiyuan.seaky@163.com)

The complicated mechanism of gas bubble oscillation in sound field is quite fascinating. One simplified case is a single bubble motivated by specific finite-amplitude sound waves. Based on the modified Keller-Miksis model, a theoretical and numerical investigation of the regular and nonlinear radial oscillations of a gas bubble driven by different given finite-amplitude waves is presented. Some crucial control parameters of the bubble radial oscillations, including radius of gas bubble, acoustic frequency and acoustic pressure, are studied with multiple numerical analysis methods. Parameter regions which are benefit for power spectrum variation of the excitation forces are given, which can provide the foundation for potential engineering applications.

9:45–10:00 Break

10:00

5aUW8. Modes of plates and shells in water driven by modulated radiation pressure of focused ultrasound. Timothy D. Daniel, Phil L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ahmad T. Abawi (HLS Res., La Jolla, CA), and Ivars Kirsteins (NUWC, Newport, RI)

We used modulated radiation pressure of focused ultrasound to excite resonant flexural vibrations of an aluminum circular plate in water. The source transducer was driven with a double-sideband suppressed carrier voltage as previously used for exciting low-frequency modes of liquid drops [P. L. Marston and R. E. Apfel, J. Acoust. Soc. Am. **67**, 27–37 (1980)]. The response of the target (detected with a hydrophone) was at twice the modulation frequency and proportional to the square of the drive voltage. Due to the spatially localized nature of the radiation pressure of the focused beam, mode shapes could be identified by scanning the source along the target while measuring the target's response. Additional measurements were done with an open-ended water-filled copper circular cylindrical shell in which resonant frequencies and mode shapes were also identified. These experiments illustrate how high-frequency focused sound can be used to identify relatively low-frequency modes of elastic objects without direct contact. [Work supported by ONR.]

10:15

5aUW9. Multi-fluid cavity underwater acoustic transducer. Yongjie Sang and Yu Lan (College of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nantong St., Nangang District, Harbin, Heilongjiang Province 150001, China, sangyongjie@126.com)

A multi- fluid cavity underwater acoustic transducer is developed. The main structures of the transducer include inner Helmholtz resonator, Janus longitudinal vibration transducer and two outer Helmholtz resonators. The vibration of Janus transducer at low frequencies is enlarged by the inner Helmholtz resonator, and the vibration of Janus transducer at high frequencies is enlarged by the two outer Helmholtz resonators, so this kind of transducer can transmit broadband acoustic signal in water. Operation bandwidth of the multi-fluid cavity underwater acoustic transducer is usually more than two octaves. The broadband operating mechanism of the transducer is studied. The effect of some of the major structural parameters on the operation bandwidth and Helmholtz resonance frequency is studied by finite element software ANSYS. The structure size of virtual prototype is obtained. A prototype is produced based on finite element design results, the electro-acoustic performance are tested in anechoic tank. The results agree well with the simulation results. Operation bandwidth of the prototype is 500 Hz–2500 Hz, the maximum source level greater than 200 dB.

10:30

5aUW10. Metal electrode direct sound generation in seawater. Michael S. McBeth (SPAWAR Systems Ctr. Atlantic, SSC Atlantic/NASA Langley Res. Ctr., 11 West Taylor St., M.S. 207, Hampton, VA 23681, m.s.mcbeth@ieee.org)

Thermoacoustic expansion of water due to Ohmic heating and volume changes in the very thin Helmholtz double layer that forms adjacent to metal electrodes are two mechanisms that have been used to explain sound generation by metal electrodes in seawater. Using 20 gauge solid sterling silver wire electrodes with only the circular cross section of the wire exposed, our experiments have demonstrated intense sound production in the form of acoustic tone bursts in seawater from a thermoacoustic mechanism at a second harmonic of 20 kHz of the electrical driving frequency at 10 kHz. Additionally, we observed a separate intense and transient acoustic tone burst generation from a mechanism we attribute to vesicle layer excitation where the vesicle is a highly charged layer of water associated with the boundary of droplets and bubbles. This vesicle layer excitation sound generation mechanism is distinct from any Helmholtz double layer mechanism since the double layer is always present while the acoustic tone burst we observe repeatedly occurs after about 4.4 ms of applied alternating voltage at 10 kHz and last only about 560 μs. We present our experimental setup and results. Moving seawater directly to generate sound may lead to more compact projectors.

10:45

5aUW11. A comparison between flexible disk transducer cavitation measurements and predictions. Arthur Horbach and James McEachern (Navmar Appl. Sci. Corp., 65 West St. Rd., Warminster, PA 18974, horbach@navmar.com)

An investigation of the threshold of cavitation of flexible disk transducers used as underwater acoustic sources was conducted. The principal task was the estimation of the cavitation threshold in sound pressure level (SPL) of an individual transducer as a function of operating depth and the actual measurement of the cavitation threshold. The transducers under test were driven at a series of voltage levels, using CW pulse lengths of 50 and 100 ms. these drive voltages were increased in steps and the received signals were analyzed for total harmonic distortion (THD), transmitting voltage response (TVR), output SPL, and voltage-current phase angle, all as a function of depth. The onset of cavitation on the flex disk is well defined in terms of THD. Comparing the measured values for the onset of cavitation to those predicted by an expression recommended by Sherman and Butler, a value of the cavitation parameter, gamma, was determined to be about 0.36, consistent with, and bracketed by, their values, for a number of transducer shapes. predictions of cavitation threshold versus depth for the flexible disk transducer were within 1 to 2 dB of measured results.

11:00

5aUW12. Spatial structure of temporal coherence scale under the different propagation conditions. Linhui Peng, Jianhui Lu, and Xiaotao Yu (Ocean Technol., Information College, Ocean Univ. of China, 238 Songling Rd., Qingdao, Shangdong 266100, China, penglh@ouc.edu.cn)

Internal wave induce the decreasing of temporal coherence of sound field. The scale of temporal coherence is related with strength of the internal wave and frequency of the sound wave; meanwhile, it has the definite spatial structure. Spatial structures of the temporal coherence are not same under the different propagation conditions. It is related with the structure of the sound field.

11:15

5aUW13. Experimental study of ultra shallow water acoustic wave propagation. Konstantin Dmitriev, Alisa Dorofeeva, and Sergei Sergeev (Phys., Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru)

The sound propagation in ultra shallow water was studied experimentally. The investigation was carried out in the nature pond approximately 300×20 m in size. Its depth was about 1 m. For such a pond critical frequency is about 700 Hz that is easy to emit and receive. The main purpose was to study the process of acoustic wave propagation in case of strong bottom influence and to restore some bottom parameters. The signal was radiated by the transducer and recorded by a set of point receivers located at different distances from the transducer. The signal was linear frequency modulated in 100 Hz–10 kHz band and was 10 s in duration. Its correlation with the recorded signal defines the pond response function. The modes representation was used. The group velocity frequency dependence was obtained by use of this method and it was very close to the theoretical one considering a soft bottom. The critical frequency also corresponded to the soft bottom model. The attenuation of sound was also studied in a wide frequency range including non-propagating mode. The results are very sensitive to the signal-to-noise ratio. It was shown that the used correlation method significantly improved the quality of obtained characteristics.

Poster paper 5aUW14 will be on display for the entire session.

5aUW14. Simulated joint reconstruction of shallow water features using acoustic tomography methods. Sergei Sergeev, Andrey Shurup, and Alisa Scherbina (Phys., Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru)

The possibility of simultaneous reconstruction of different shallow water parameters by the mode tomography methods is considered. The initial data for the reconstruction are the propagation times of individual modes in the different frequency bands. The reconstruction results can be both the water layer parameters (such as sound speed profile, eddies) and the bottom characteristics (such as relief and sound speed in the bottom). The approach is based on idea that the different shallow water parameters affect the propagation times of different modes on the different frequencies in the different manner. As a result, if it is possible to separate from the received total field enough information about the propagation times of the mode signals, then it becomes possible to implement the joint reconstruction of the considered shallow water features. Another peculiarity of the regarded approach that all the considered shallow water features are described using the single basis. As a result, the single perturbation matrix is used for the joint reconstruction. The numerically simulated results obtained by the regarded approach are presented.