Session 3pAA

Architectural Acoustics: American Institute of Architects Continuing Education Units Course Presenter Qualification

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

Bennett M. Brooks, Cochair
Brooks Acoustics Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066

Norman H. Philipp, Cochair
Geiler and Associates, LLC, 1840 E. 153rd. Cir., Olathe, KS 66062

Chair's Introduction—1:00

Invited Papers

1:05

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

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

2:05

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

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

Session 3pAB

Animal Bioacoustics, Signal Processing in Acoustics, Psychological and Physiological Acoustics, and Speech Communication: Neural Mechanisms of Complex Sound Discrimination II

Andrea Simmons, Cochair
Brown Univ., Box 1821, Providence, RI 02912

Hiroshi Riquimaroux, Cochair
Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan

Contributed Papers

1:00

3pAB1. Timing patterns of strobe groups for echolocating big brown bats performing a target detection task. Laura N. Kloepper, James A. Simmons, Jason E. Gaudette (Dept. Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, laura_kloepper@brown.edu), Ryan Himmelwright, and Dan Robitzski (Neurosci., Lafayette College, Easton, PA)

While flying in dense clutter or close to objects, bats often produce “strobe groups,” pairs of pulses emitted at short pulse intervals followed by longer pulse intervals. Previous studies of free-flying bats demonstrate relatively consistent trends in strobe groups depending on the degree of clutter. To investigate strobe group production in stationary bats, three big brown bats were trained to perform a target detection task and their echolocation signals were analyzed. Variation in the number of pulses within strobe groups as well as strobe group characteristics varied substantially between individuals and experimental trials, yet when strobe groups were produced the time between pulses and strobe groups remained relatively stable. These data suggest that although bats demonstrate flexibility in the production and characteristics of strobe groups, the inherent timing of pulses within and across strobe groups is stable. These patterns may reflect precise timing adaptations in sound-producing motor circuits of individual bats.

1:15

3pAB2. The role of saccular resonance in fish audition. Mardi C. Hastings and Rachel Rozin (Georgia Inst. of Technol., George W. Woodruff School of Mech. Eng., 801 Ferst Dr., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

Dynamic characteristics and resonance of the saccule play a fundamental role in audition in all teleosts, including those with direct connections between the swim bladder and inner ear. The saccule is an accelerometer with its rigid mass, the otolith (or saggita), coupled to the sensory epithelium through mechanical impedances of the otolithic membrane and hair-cell ciliary bundles. Relative displacement between the saggita and sensory epithelium induced by sound correlates with hearing sensitivity. Dynamic models of the peripheral auditory system in fishes from five different orders (the oscar, broad whitefish, oyster toadfish, dab, and goldfish) were developed for a comparative analysis. Species selected included one without a swim bladder and one with Weberian apparatus that transmits swim bladder motion directly to the saccule. Results for all fishes agreed with audiograms published in the literature. The lowest frequency marking the band of best sensitivity was found to be at the saccular resonance in all species; however, width of the band depended on excitation of the saccule indirectly from motion of the swim bladder and/or Weberian apparatus. Species with swim bladders and larger saggita had best sensitivities at lower frequencies, but with smaller bandwidths because the saccule could not respond to indirect stimulation.

1:30

3pAB3. A model for peripheral auditory mechanics in the oyster toadfish, *Opsanus tau*. Rachel Rozin (Georgia Inst. of Technol., Georgia Inst. of Technol., Atlanta, GA 30332-0405, rachelrozin@comcast.net), Peggy L. Edds-Walton (Woods Hole Oceanogr. Inst., Woods Hole, MA), and Mardi C. Hastings (Georgia Inst. of Technol., Atlanta, GA)

Frequency response of the peripheral auditory system in the oyster toadfish was analyzed using a biomechanical model based on morphometric data obtained from a CT scan of a mature female, 21-cm long. Tissue properties for system equations were estimated from those found in the literature. The saccule is considered to have a single degree-of-freedom corresponding to the primary directional orientation of the hair cells. The model determines relative displacement between the sensory epithelium and otolith due to response of the saccule to motion from the sound source (direct path) and swim bladder (indirect path). Largest relative displacements correlate with highest auditory sensitivity (lowest thresholds). Results indicate a flat response at low frequencies with high sensitivity near 100 Hz, and are in good agreement with best stimulus frequencies measured physiologically in the auditory medulla and midbrain, and with the toadfish behavioral audiogram. Moreover, results confirm that the indirect path has little, if any, influence on auditory thresholds. Detection of the phase difference between direct and indirect signals, however, may contribute to the ability of oyster toadfish to localize sound sources.

1:45

3pAB4. Auditory steady-state response measurement of the temporal dynamics of hearing sensitivity in an echolocating bottlenose dolphin (*Tursiops truncatus*). Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmpfoundation.org), James J. Finneran (US Navy Marine Mammal Program, SSC Pacific Code 71510, San Diego, CA), and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

Studies with some echolocating odontocetes demonstrate that receiver-based automatic gain control (AGC) compensates for reductions in echo strength resulting from acoustic spreading loss. This study examined AGC in an echolocating bottlenose dolphin by measuring changes in hearing sensitivity over time courses corresponding to single click-echo pairs. The electrophysiological auditory steady-state response (ASSR) elicited by a 113-kHz sinusoidally amplitude-modulated tone was recorded while the dolphin performed a target discrimination task. Auditory electrophysiological responses were extracted from the instantaneous electroencephalogram and coherently averaged using the modulation rate of the 113-kHz tone as a reference. A Fourier transform was then performed with a 10-ms sliding window to obtain the ASSR amplitude as a function of time relative to the dolphin’s outgoing click and received echo. The ASSR amplitude initially decreased at the time of click emission and then recovered over a course of 25 to 70 ms, depending on target range. This relatively long time course of recovery appears to be consistent with forward-masking, as opposed to an AGC mechanism based on the contraction and gradual release of middle ear muscles coincident with click emission. [Work funded by SSC Pacific Naval Innovative Science and Engineering (NISE) program.]

2:00

3pAB5. Investigating biosonar automatic gain control in a dolphin using auditory evoked potentials. James J. Finneran (US Navy Marine Mammal Program, SSC Pacific Code 71510, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow, and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

Studies with echolocating odontocetes suggest that forms of automatic gain control mediate auditory electrophysiological responses to target echoes. This study used a phantom echo generator and auditory evoked potential measurements to examine automatic gain control in a bottlenose dolphin. Auditory evoked potentials to outgoing clicks and incoming echoes were recorded for simulated ranges from 2.5 to 80 m. When geometric spreading loss was simulated, echo-evoked potential amplitudes were essentially constant up to 14 m and progressively decreased with increasing range. When the echo levels were held constant relative to clicks, echo-evoked potential amplitudes increased with increasing range up to 80 m. These results suggest that automatic gain control maintains distance-independent echo-evoked potential amplitudes at close range, but does not fully compensate for attenuation due to spreading loss at longer ranges. The automatic gain control process appears to arise from an interaction of transmitter and receiver based processes, resulting in a short-range region of distance-independent echo-evoked potential amplitudes for relevant targets, and a longer-range region in which echo-evoked potential amplitudes are reduced. [Work funded by SSC Pacific Naval Innovative Science and Engineering (NISE) program.]

2:15

3pAB6. Not-so-automatic gain control in the bottlenose dolphin: Source level distance compensation depends on prior knowledge of target distance. Laura N. Kloepper (Dept. Neurosci., Brown Univ., 185 Meeting St., Box GL-N,Providence, RI 02912, laura_kloepper@brown.edu), Paul E. Nachtigall, and Adam B. Smith (Zoology, Univ. of Hawaii, Honolulu, HI)

Although termed “automatic gain control,” previous field and laboratory investigations into source level distance compensation in bats and odontocetes have relied on animals echolocating targets or arrays at predictable distances. To test the “automatic” nature of gain control in the bottlenose dolphin, the source level distance compensation was measured for three target distances (2, 4, and 7 m) in two types of sessions: predictable, in which the target distance was held constant within a session, and random, in which the target distance varied within a session. In the predicable sessions the dolphin demonstrated source level distance compensation at a rate of 10 log (distance), a level approximately half that predicted by past gain control experiments. In the random sessions the dolphin did not demonstrate source level distance compensation and, regardless of target distance, produced source levels that were equivalent to those produced for the predictable sessions at 4 m distance. These data suggest that gain control is not automatic, and, in the absence of prior knowledge of target distance, echolocating animals may adopt a strategy of fixing their source level to an intermediate distance of predicted target range.

WEDNESDAY AFTERNOON, 4 DECEMBER 2013 GOLDEN GATE 2/3, 1:00 P.M. TO 3:15 P.M.

Session 3pBA

Biomedical Acoustics: General Topics in Biomedical Acoustics

Robert McGough, Chair
Dept. of Elec. and Comput. Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48824

Contributed Papers

1:00

3pBA1. A technique for measuring bone density using an ultrasonic imaging system. Catherine J. Miller, Morgan R. Smathers, Cameron R. Thurston, and Brent K. Hoffmeister (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, milcj-16@rhodes.edu)

Introduction: Osteoporosis is a degenerative bone disease that affects millions of Americans. Osteoporosis causes normally porous bone tissue, called cancellous bone, to become more porous and weak. It is possible that ultrasonic imaging systems may be used to detect changes in bone density (porosity) caused by osteoporosis. Methods: Ultrasonic images were acquired from 25 cube shaped specimens of cancellous bone in a water tank using a Terason 2000 + ultrasonic imaging system with a 5 MHz linear array transducer. Images were analyzed using an image processing program called ImageJ. Pixel brightness values were plotted as a function of depth in the images of each bone specimen. Pixel value gradient (PVG) was defined as the slope of the resulting graph. Results: PVG was negative for all specimens, and was found to decrease (become more negative) with bone density. PVG demonstrated a moderate but highly significant ($p < 0.001$) linear correlation with bone density (R = -0.79). Conclusion: Ultrasonic images of bone may be analyzed in ways that yield quantitative information about bone density.

1:15

3pBA2. Dual gate ultrasonic backscatter technique compared to x-ray microtomography parameters. Morgan Smathers, Joseph A. McPherson, Mark Sellers, and Brent K. Hoffmeister (Phys., Rhodes College, 2000 N Parkway, Memphis, TN 38112, smamr@rhodes.edu)

Over 52 million Americans suffer low bone mass and at least 10 million suffer from osteoporosis. This study seeks to develop a dual gate ultrasonic technique for predicting bone quality as well as bone quantity. Ultrasonic pulses from a 5 MHz transducer were propagated into regions of porous bone in 18 bone specimens from one bovine and four human donors. The dual gate technique considered the normalized mean of the backscatter difference (nMBD), which is the power difference between two gated regions of 2 μs each placed 1 μs apart over the returned signal. This ultrasonic parameter was compared to eight X-Ray MicroCT parameters describing bone quality and quantity. Among these are the Structural Model Index (SMI) and Relative Bone Volume (BV/TV). SMI grades the structure of a specimen based on its plate and rod characteristics, making it a bone quality characteristic. SMI produced an R value of 0.982 with nMBD. BV/TV, a bone quantity indicator, finds the ratio of bone volume in the total specimen volume, and showed an R value of 0.993 with nMBD.

1:30

3pBA3. Nonlinear propagation effects on measurement of backscatter coefficient of tissue-mimicking materials. Timothy Stiles (Phys., Monmouth College, 700 E Broadway Ave., Monmouth, IL 61462, tstiles@monmouthcollege.edu)

Measurement of the ultrasonic backscatter coefficient (BSC) holds great promise in providing quantitative diagnostic information on various diseases. Many clinical studies utilize the reference phantom method. In this method, the measured BSC is the ratio of the power spectrum of the scattered sound from the patient to the power spectrum from the reference phantom multiplied by the known BSC of the reference and an attenuation correction factor. In these studies, nonlinear propagation has been ignored. Nonlinear propagation causes changes in the power spectrum as incident energy is converted to harmonics and the acoustic signal undergoes nonlinear attenuation. This study characterized the effects of nonlinear propagation on reference phantom measurements of BSC from four tissue-mimicking samples in the frequency range from 2 to 20 MHz and incident pressure from 1 to 10 MPa (measured in water) using single element focused transducers. Samples consisted of glass microspheres suspended in a mixture of agar and concentrated milk. The resulting BSC varied by up to a factor of 30 depending on the incident pressure. Substantial changes in the overall shape of the BSC vs frequency were also observed. New methods of measurement of BSC that account for nonlinear propagation are explored.

1:45

3pBA4. 20—100 kHz, ultrasound assisted treatment of chronic wounds. Peter A. Lewin, Joshua A. Samuels (Biomed_7_701, Drexel Univ., 3145 Market St., Philadelphia, PA 19104, plewin@coe.drexel.edu), Michael S. Weingarten (Dept. of Surgery, Drexel Univ. College of Medicine, Philadelphia, PA), Leonid A. Zubkov, Youhan Sunny, Christopher R. Bawiec (Biomed_7_701, Drexel Univ., Philadelphia, PA), and David J. Margolis (Dept. of Epidemiology, Univ. of Pennsylvania Perelman School of Medicine,, Philadelphia, PA)

We report the results of a limited (20 patients) clinical study involving treatment of chronic wounds (venous ulcers) using novel, fully wearable ultrasound array applicator operating in the range of 20—100 kHz and generating pressure amplitudes close to 55 kPa (about 100 mW/cm^2, Sptp). 20 kHz, 15 min exposure was determined to be the most effective in terms of expediting reduction in wound area ($p < 0.05$). The applicator was designed as compact, tether-free, device that can be comfortably worn by subjects at home, permitting active (combined with traditional compression) therapy away from the clinical setting. The system is safe for extended periods of application and the arrangement of the piezoelectric elements in the applicator has been adjusted allowing patient treatment customization, especially treatment of irregularly shaped ulcers. A set of systematic experiments *in vitro* aiming at the identification of potential mechanisms of the wound healing was also performed; the results verified that the exposure matrix used in clinical setting as mentioned above (20 kHz treatment for 15 min) produced the greatest increase in cellular metabolism ($p < 0.05$) and cell (fibroblasts) proliferation ($p < 0.01$) vs. a sham. The subsequent *in vitro* experiments will focus on collagen production, crucial to wound healing. [NIH 5R01EB009670.]

2:00

3pBA5. An ultrasound technique for wireless power transmission through tissue to implanted medical devices. Leon Radziemski (Piezo Energy Technologies, 5153 N Via Velazquez, Tucson, AZ 85750, ljrpet@comcast.net) and Inder R. Makin (Piezo Energy Technologies, Mesa, AZ)

An ultrasound electrical recharging system (USER^{TM}) is developed and tested, which wirelessly transmits significant amount of energy through animal tissue to charge implantable devices, batteries, or capacitors. The goal of this approach is wireless power transmission to active human implant devices. Experiments with transducers with resonant frequencies between 0.5 and 3.5 MHz led us to adopt 0.75 to 1.25 MHz as the range of optimum efficiency. *In vitro* experiments demonstrated significant charging of 4.1 V medically qualified Li-ion batteries across tissue depths of up to 5 cm. Charging currents close to 300 mA were achieved *in vitro*. Several *in vivo* tests confirmed the power delivery in a porcine model. In an *in vivo* survival test, tissue was exposed to 1 MHz ultrasound at an average intensity of 0.4 W/cm^2 for 11.5 h. Histology of the exposed tissue showed tissue changes primarily attributable only to surgical implantation of the prototype device. Many traditional and developing implanted medical devices are targets for the introduction of this method of power delivery, to reduce the number of battery replacement operations, and improve performance compared to the existing electromagnetic method of wireless power delivery. [Work supported by the NIH/NIBIB R44EB007421.]

2:15

3pBA6. Integrated transmission-reflection quantitative ultrasound noninvasive prediction of trabecular bone principal structural orientation validated with mechanical testing. Liangjun Lin, Wei Lin, and Yi-Xian Qin (Biomedical Eng., Stony Brook Univ., 100 Nicolls Rd., Rm 212, Bio-Eng. Bldg., Stony Brook, NY 11794-5281, john85726@gmail.com)

Quantitative ultrasound (QUS) measurement was shown to have the ability to predict the principal structural orientation (PSO) with a spherical bone model. It is hypothesized that with a cubic bone model, integrated transmission-reflection QUS measurement can predict the PSO and therefore improved the correlation with the mechanical and structural parameters. Twelve trabecular bone cubes were harvested from bovine distal femur. Compression testing and μCT of 30 μm resolution were performed to obtain the mechanical and structural parameters. QUS measurement was performed on the transverse plane in a range of angles, from −-30° to 30° to medial-lateral orientation at 5° increment. For each angle, reflection mode was used to measure the thickness of the sample in the specific scanning angle. Then, the sample thickness was used to normalize the transmission mode measurement in the same angle. The thickness measured by reflection mode QUS was highly correlated to the results measured by caliber (slope = 0.99, R2 = 0.85). Compared to the traditional transmission mode, the correlation coefficients (R2) between transmission-reflection mode ultrasound velocity versus mechanical and structural parameters were improved (elastic modulus, 0.62 to 0.73; SMI, 0.74 to 0.90; BV/TV, 0.75 to 0.85; Tb.N, 0.60 to 0.75; Tb.Sp, 0.66 to 0.73).

2:30

3pBA7. A comprehensive computational model of sound transmission through the porcine lung. Ying Peng (Mech. Eng., Univ. of Illinois at Chicago, 2951 S King Dr. Apt. 1805, MIE, Chicago, IL 60616, YPENG6@UIC.EDU), Zoujun Dai, Brian Henry (BioEng., Univ. of Illinois at Chicago, Chicago, IL), Hansen Mansy (Rush Univ., Chicago, IL), and Thomas Royston (BioEng., Univ. of Illinois at Chicago, Chicago, IL)

A comprehensive computational simulation model of sound transmission through the porcine airways and lung is introduced and experimentally evaluated. This "subject-specific" model utilizes parenchymal and major airway geometry derived from x-ray CT images. The lung parenchyma is modeled as a poroviscoelastic material using Biot theory. A finite element (FE) mesh of the lung that includes airway detail is created, and COMSOL FE software is used to simulate the vibroacoustic response of the lung to sound input at the trachea. The FE model is validated by comparing simulation results with experimental measurements using scanning laser Doppler vibrometry on the surface of an excised and preserved lung. The FE model is also used to calculate and visualize vibroacoustic pressure and motion inside the lung and its airways caused by the acoustic input. The effect of diffuse lung fibrosis and a tumor on the lung acoustic response is simulated and visualized using the FE simulation. In the future, this type of visualization can be compared and matched with experimentally-obtained elastographic images in order to better quantify lung material properties. [Work supported by NIH Grant EB012142.]

2:45

3pBA8. A simulation model of cyclic variation of red blood cell aggregation under Couette and Poiseuille flows. Qi Kong, Kwon-Ho Nam, and Dong-Guk Paeng (Ocean system Eng., Jeju National Univ., Rm. 5464,College of Ocean Sci. 4th Bldg., Jeju National Univ., Ara 1-Dong Jeju-Si Jeju-Do, Jeju, South Korea, kongqi2011@gmail.com)

The aggregation of red blood cells (RBCs) is a reversible phenomenon in which RBCs form a pile or network at low shear rate via the interactions of electrostatic and aggregating forces. In previous experimental results under both Couette and Poiseuille flows, cyclic variations in blood echogenicity were observed but their cyclic patterns were different. In this paper, a two-dimensional particle model capable of mimicking the main characteristics of RBC aggregation kinetics was proposed to elucidate the different effects of hemodynamics under Couette and Poiseuille flows on RBC aggregation. In simulation results, cyclic variation of RBC aggregation was observed but its magnitude of mean aggregate size (MAS) was not changed by variations of velocity and stroke rate under Couette flow. These results are in agreement with the experiment results. In contrast, the simulation results under Poiseuille flow revealed that cyclic variation of RBC aggregation and its MAS magnitude were changed by variations of velocity and stroke rate. As stroke rate increased from 20 to 120 beats/min, the phase of MAS variation compared with velocity profile was shifted. These simulation results may provide the theoretical explanation of the different experimental results of cyclic variation of RBC aggregation under Couette and Poiseuille flows. [Work supported by NIPA-2013-H0401-13-1007 and 2013R1A1A2043478.]

3:00

3pBA9. Errors in ultrasonic scatterer size estimates due to mixed scatterer populations. Anthony L. Gerig and Breanna P. Swan (Mathematics and Phys., Viterbo Univ., 900 Viterbo Dr., La Crosse, WI 54601, algerig@outlook.com)

Ultrasonic scatterer size estimation provides an accurate measure of actual scatterer size when those sizes are narrowly distributed about a single, mean value. Although often the case, there are instances in tissue where two or more scatterer types with significantly different sizes are believed to contribute to the same signal. The purpose of this work is to characterize the errors in the size estimates obtained for one scatterer type when contaminating scatterers of a second type are present. Theoretical results for the error are compared with simulation and experimental results for uniform phantoms containing a binary mixture of scatterers. These results indicate that errors can be significant for the frequency bands typically used in size estimation, especially when contaminant scatterers are larger than the scatterers of interest. Results also indicate that, however, these errors can be reduced by shifting the frequency band used to estimate size. A technique for correcting the errors is also described and applied to the phantom data. Although effective, the method requires prior knowledge of the backscatter coefficient of the contaminant scatterers, and the variability of the corrected values can limit its utility as contaminant scattering strength increases.

WEDNESDAY AFTERNOON, 4 DECEMBER 2013 MASON, 1:30 P.M. TO 2:45 P.M.

Session 3pEA

Engineering Acoustics: Non-Traditional Electro-Acoustic Transducer Design II: Contemporary Micro-Mechanical Devices

John B. Blottman, Chair
Div. Newport, Naval Undersea Warfare Ctr., 1176 Howell St., Code 1535 B1170/108, Newport, RI 02840

Contributed Papers

1:30

3pEA1. Characterization of a microelectromechanical microphone using the finite element method. Mads J. H. Jennings (Comsol AB, Kgs. Lyngby, Denmark), Wade Conklin, and Jordan Schultz (Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60143, wade.conklin@knowles.com)

Recent development and advances within numerical techniques and computers now enable the modeling, design, and optimization of many transducers using virtual prototypes. Here, we present such a virtual prototype of a Knowles SiSonicTM MEMS microphone. The virtual prototype is implemented using the finite element method with COMSOL Multiphysics and includes description of the electric, mechanic, and acoustic properties of the transducer. The acoustic description includes thermal and viscous losses explicitly solving the linearized continuity, Navier-Stokes, and energy equations, that is, thermoacoustics. The mechanics of the diaphragm are also modeled including electrostatic attraction forces and acoustic loads. A sub-model approach is used to model and lump the acoustic properties of the small perforations in the microphone backplate. The model has no free fitting parameters and results in the prediction of the frequency response, in the audible and ultrasound range, as well as other relevant characteristics. The model results show good agreement with measured data.

1:45

3pEA2. A piezoelectric micromachined ultrasonic transducer array for parametric loudspeakers. Yub Je (Mech. Eng., Pohang Univ. of Sci. and Technol., San 31, Hyojadong Namgu, Pohang, Kyungbuk, KIRO 416, Pohang, Gyungbuk, South Korea, wkmoon@postech.ac.kr), Haksue Lee (Agency for Defense Development, Jinhae, South Korea), and Wonkyu Moon (Mech. Eng., Pohang Univ. of Sci. and Technol., Pohang, South Korea)

A parametric array is a nonlinear conversion process that can generate a highly directional sound beam with a small aperture. Since parametric sound generation requires high-intensity ultrasound for nonlinear interaction, efficient sound generation is an important issue in the practical use of parametric loudspeakers. In this study, a piezoelectric micromachined ultrasonic transducer array was investigated in order to generate directional audible sound with a parametric array. A piezoelectric micromachined ultrasonic transducer, with a micron-thick radiating plate, was verified to generate ultrasound with high-efficiency in air. Two types of unit transducers with different resonance frequencies (f1 = 100 kHz, and f2 = 110 kHz) were arranged in the transducer array to extend the frequency bandwidth. The electroacoustic efficiency of the transducer was measured to 71% at its resonance frequency. The $\pm$3dB-frequency bandwidth of the transducer array was 17 kHz. The spatial distributions of the difference frequency wave were measured and compared with the computed data in the audible frequency range. The fabricated transducer array consumed 1 W of electric power while generating a 10 kHz-difference frequency wave with sound pressure level of 80 dB.

2:00

3pEA3. The performance enhancement of a micro-machined microphone based on field-effect-transistor and electrets. Kumjae Shin, Yub Je (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technol., KIRO 416 Hyoja-Dong, Nam-Gu, Pohang, Gyungbuk 790-784, Korea, Republic of, forhim13@postech.ac.kr), Haksue Lee (6th R&D Institute-1, Agency for Defence Development, Changwon, Kyoungnam, South Korea), James E. West (Dept. of Electric & Comput. Eng., Johns Hopkins Univ., Baltimore, MD), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technol., Pohang, Gyungbuk, South Korea)

Most microphones use a capacitive type transduction. However, this form of transduction faces problems with microminiaturization. The most prominent issue is a decrease in sensitivity at low frequency. Although several works suggested a microphone based on the field-effect-transistor (FET) to solve this problem, other issues, such as low signal to noise ratio and a need for high bias voltage due to metal electrode, remained. To overcome this limit, a micro-machined microphone based on the FET and electret was proposed, and its feasibility was shown in 2012. Its principle is that the electric field arising from the electret controls the channel of the FET embedded on the membrane. Although its feasibility as an acoustic-sensitive device was shown, several problems still exist in terms of stability, sensitivity, and noise. To realize stable and highly sensitive modulation, parametric analysis for the transduction was done to enhance performance. In particular, the surface potential of the electret was increased more than the previous one. It resulted in a stronger electric field applied at the gate. Therefore, it made the FET more sensitive to membrane vibration. The sensitivity evaluation setup was modified for a more accurate measurement. The evaluation results are to be presented.

2:15

3pEA4. Experimental comparison of 3-3 and 3-1 mode piezoelectric microelectromechanical systems. Donghwan Kim (Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712, donghwan.kim@utexas.edu), Nishshanka N. Hewa-Kasakarage, Michael L. Kuntzman, and Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

A common architecture for piezoelectric MEMS sensors and actuators is a thin piezoelectric film patterned atop a much thicker passive bending structure (e.g., a silicon beam or plate). In a first common configuration, parallel plate electrodes reside above and below the piezoelectric film to realize a 3-1 mode device. In a second configuration, a top electrode is patterned in the form of an interdigitated transducer (IDT) to realize a parallel network of 3-3 mode cells. A theoretical comparison of figures-of-merit for each configuration has been presented by research teams in the past. Figures-of-merit include coupling coefficient, actuator strength, and signal to noise ratio for sensing applications. Less work has been performed directly comparing these configurations experimentally using micro-processed thin films. In this presentation, a micromachined accelerometer structure employing a set of multiple springs is used to experimentally compare the two configurations. Each silicon spring contains a 1micron thick lead zicronate titanate (PZT) film along the top surface. Aside from electrode type, the springs are identical in dimension—providing an opportunity for direct comparison.

2:30

3pEA5. A micro-machined Tonpilz hydrophone for audible frequency sounds. Min Sung (Mech. Eng., POSTECH, PIRO416, San31, Hyoja, Namgu, Pohang city, Kyungbuk, Pohang, Kyungbuk 790784, South Korea, smmath2@postech.ac.kr), Haksue Lee (Underwater Acoust. Lab., Agency for Defense Development, Changwon, Kyungnam, South Korea), and Wonkyu Moon (Mech. Eng., POSTECH, Pohang, Kyungbuk, South Korea)

A micro-machined Tonpilz hydrophone based on the piezoelectric thickness mode was designed for audible frequency range of 20 Hz~20kHz. The structure of the sensor was motivated by conventional Tonpilz transducers, but two different design approaches were adopted to enhance the sensitivity and to endure the high hydrostatic pressure in deep-sea. For improved sensitivity, the area ratio of the head-mass and the piezoelectric body was designed to be several hundreds to one, which amplifies the input of the transduction body due to acoustic pressure. Since this approach is adopted in order to develop a miniaturized hydrophone manufactured by a batch process, the size of the piezoelectric transduction body becomes too small to generate the amount of charge enough for detecting the signal using the conventional pre-amplifier at low frequencies below 500 Hz. We have developed and validated the lumped parameter model and used it to identify the requirements for the pre-amplifier circuits and the available sensitivity at low frequencies and to search for the appropriate design for miniaturized hydrophone. Then, the designed hydrophone was fabricated by micromachining and assembled with the custom-made pre-amplifier inside with caster-oil filled rubber housing. The evaluation of the hydrophone will be presented. [Research supported by MRCnD.]

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Natalia Sidorovskaia, Chair
Dept. of Phys., Univ. of Louisiana at Lafayette, UL BOX 44210, Lafayette, LA 70504-4210

Chair's Introduction—2:00

Invited Paper

2:05

3pED1. Time-frequency analysis for acoustics education and for listening to whales in the Gulf of Mexico. Juliette W. Ioup (Dept of Phys., Univ. of New Orleans, New Orleans, LA 70148, jioup@uno.edu)

Time-frequency plots continue to be used in many varied applications. One particularly advantageous use is in acoustics courses accessible to non-science majors, students who are often frightened of mathematics and/or physics. All musicians as well as many others can read and understand music scores (time-frequency plots). Time-frequency plots are extremely useful in explaining the differences in timbre of the same pitch coming from different musical instrument families, from individual instruments themselves, and from different human voices. Examples are given from the first of a UNO two-semester sequence on the Physics of Music (textbook by Rossing!). The second semester of this sequence includes recording and reproduction of music, and time-frequency plots are again very useful. Investigation of acoustic signals for research also benefits from the use of time-frequency plots. The study of marine mammals is enhanced by analysis of underwater acoustic recordings. Examples of both the sounds of and the time-frequency plots for sperm whale clicks in the northern Gulf of Mexico are presented. Seismic airgun shots from oil industry exploration can be heard on the recordings as well as the whale clicks.

WEDNESDAY AFTERNOON, 4 DECEMBER 2013 CONTINENTAL 6, 1:15 P.M. TO 3:00 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Frederick J. Gallun, Chair
National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239

Chair's Introduction—1:15

Invited Papers

1:20

3pID1. Hot topics in architectural acoustics: Classroom acoustics and archaeoacoustics. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514, dlubman@dlacoustics.com)

When ASA members discovered that typical American classrooms were too noisy or reverberant for serious learning in 1988 they began a successful grassroots movement to fix them. By 2002, ASA volunteers produced the first-ever ANSI standard for classroom acoustics. The effort was led by ASA's TCAA and supported by three other TCs, the S12 Standards Committee, courageous ASA staffers, and elected officers. ANSI Standard S12.60 was adopted fully or partly by school districts, states, and architectural authorities including the Green Building Council. Classroom acoustics research reporting remains active at ASA meetings. We show how this standard helps to make a better world. The new fields of archaeoacoustics and historical acoustics are "hot". They employ scientific

acoustics to study the past. Their novel hypotheses and discoveries attract young investigators to acoustical careers. Sound was more important in the quiet ancient world. Many are fascinated by the 1988 discovery of strong correlation between the locations of Paleolithic cave paintings and cave resonance. Why? A pyramid at Chichen Itza, Mexico, chirps like a bird revered in Mesoamerican cultures. The chirp is explained by applying the convolution theorem to pyramid architecture. Was it intentionally designed? Other archaeoacoustic and historical acoustic examples are addressed.

1:45

3pID2. Single beam acoustic tweezer. K. K. Shung (Biomed. Eng., Univ of S. Calif., 136 DRB, 1042 Downey Way, Los Angeles, CA 90089, kkshung@usc.edu)

Single beam acoustic tweezer, a distant cousin of optical tweezer, has been recently experimentally validated. A prerequisite of acoustic tweezer as in optical tweezer is a sharply focused beam with a steep intensity variation within the dimension of a particle. As the frequency of an acoustic beam reaches 100 MHz or higher, the beam diameter approaches cellular level allowing acoustic tweezing or trapping of a cell. Recent experimental results have shown that at 200 MHz it is possible to trap particles as small as 1 μm and red blood cells (RBC). These results along with the experimental arrangement and potential biomedical applications of acoustic tweezer including measuring RBC deformability will be discussed in detail in this talk.

2:10

3pID3. Hot topics in musical acoustics applied to real-time sound synthesis. Julius O. Smith (Music / CCRMA, Stanford Univ., Stanford, CA 94305, AbstractCentral@w3k.org)

(Invited paper for an Interdisciplinary session) New activities in any field are often precipitated by new enabling technologies. In musical acoustics applied to real-time sound synthesis, one exciting new development is smart-phones and tablets having high audio quality, multicore processing power, and display screens with multitouch controls. For example, it is possible to implement a complete virtual-acoustic guitar on current smart-phones and tablets, playable in real time, with plenty of processing power left over for real-time display of chord charts and vibrating-string graphics. (See presentations of the moForte Guitar at this conference and/or on the Web.) A second enabling technology area is evolving high-level domain-specific languages for audio signal processing such as Functional AUdio STream (FAUST). For example, the source code for the moForte virtual acoustic guitar, written in the FAUST language, is about an order of magnitude smaller (and faster to write) than conventional C++, while the compiled performance is generally within a factor of 2. A third enabling technology area is advancement in methods for convex optimization along with advances in the development of convex formulations of important problems. For example, a convex formulation has recently been developed for approximating 2D bowed-string bridge admittances as "passive" (positive-real) digital filters. (See presentations by Esteban Maestre.)

2:35

3pID4. Hot topics: Recent discoveries using new recording methods in animal bioacoustics. James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu)

Recent research has successfully addressed a variety of difficult questions about animal acoustic communication and orientation by taking advantage of two related techniques—on-board recording or telemetry of animal sounds or neurophysiological signals during behavior, and acoustic tracking of the locations of animals using arrays of microphones or hydrophones coupled with efficient software for processing the time-of-arrival of sounds. These methods have great power for relating the locations and movements of animals to their acoustic signals and auditory responses, but they have limitations that have to be taken into account if their potential is to be realized. The examples to be presented highlight discoveries that have been made using these methods. They have broadened our understanding of how animals interact acoustically with each other according to their spatial distribution and proximity, how they dynamically regulate auditory sensitivity.

Session 3pNS

Noise and Architectural Acoustics: Double Duty: The Added Value of Coupling Acoustics with Other Functionality

David S. Woolworth, Chair
Oxford Acoust., 356 CR 102, Oxford, MS 38655

Chair's Introduction—1:25

Invited Papers

1:30

3pNS1. Supporting dynamic infrastructure loads from a vibrating structure. Alexander Salter (Charles M. Salter Associates, Inc., 130 Sutter St., Fifth Fl., San Francisco, CA 94104, alex.salter@cmsalter.com)

A large ballroom at a downtown hotel in San Francisco sits directly below the parking garage with ramps located both under and over the Ballroom's parlor areas on either side. In 2011 Charles Salter Associates, Inc., together with the project architect and structural engineer, developed an isolated structural grid to support the ballroom infrastructure. The grid also allows for variable loading at select rigging points to accommodate a wide range of events. Prior to the renovation, significant vibration and structure borne noise was generated inside the ballroom due to vehicular activities. The ceiling structure, ductwork, and piping, as well as the chandeliers visibly shook and rattled during vehicle pass-bys. As a result, the hotel received complaints and needed to mitigate the issue in order to provide a first class ballroom facility. This paper discusses the background and design process taken in order to mitigate the issue, and describes how the isolated grid provides both acoustical mitigation as well as utility to users of the hotel's ballroom.

1:50

3pNS2. Combining acoustical and fire-resistive design in separating assemblies. John LoVerde and David W. Dong (Veneklasen Associates, 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

In most project types, separating assemblies must be fire rated, which means the assembly design and materials must be listed by one of the approved fire rating bodies. This can limit the acoustical design options to assemblies and materials that have previously been listed. For large developers, it can be advantageous to design new assemblies that are not currently listed that meet both the acoustical goals and the fire code requirements. Acoustical testing is performed on design iterations and with various products, in coordination with architectural elements and product manufacturers. The completed assembly is burn testing and agency listed. Examples and lessons of the process are shared.

2:10

3pNS3. The importance of the relationship of acoustics and sustainability. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

Sustainability has become a keyword in the architecture and the building industry, in no small part driven by the development of standards such as IGBC and LEED. The refined definition of sustainability, however, depends on the viewpoint: economic, social, or environmental. Acoustics of buildings (and the environment) has an intrinsic link to sustainability that should be highlighted to prevent the sidelining of architectural acoustics in building budgets and architectural programs of study. This paper examines the concept of sustainability as it relates to the building industry and architectural acoustics, and provides a summary of work in this area to outline this relationship to illuminate the importance of acoustics in sustainability.

2:30

3pNS4. Relating acoustics and thermal performance. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

Good thermal performance of a building produces a tangible financial return to the owner that comes from energy savings. While some not-as-directly tangible returns such as functionality or productivity can be exacted from good acoustics, some acoustic treatment materials or techniques also exhibit beneficial thermal properties. This paper examines and quantifies some of these acoustic approaches that allow thermal benefits to "piggy back" and help embed the acoustic design into the project so that it is not so easily removed in the value engineering phase of the project.

Session 3pPAa

Physical Acoustics, Noise, Structural Acoustics and Vibration, and Engineering Acoustics: Jet and Other Aeroacoustic Noise Source Characterization II

Kent L. Gee, Cochair
Brigham Young Univ., N243 ESC, Provo, UT 84602

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

Invited Papers

1:00

3pPAa1. Sensitivity analysis of an equivalent source model for a military jet aircraft. Tracianne B. Neilsen, Kent L. Gee, David M. Hart (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), and Michael M. James (Blue Ridge Research and Consulting, LLC, Asheville, TN)

The noise from a tied-down F-22A Raptor is modeled with an equivalent source consisting of two line arrays of monopole sources and their image sources, to represent the interference from the ground. These arrays, one correlated and one uncorrelated, with Rayleigh-distributed amplitudes, mimic properties of fine and large-scale turbulent mixing noise. [Morgan *et al.*, Noise Control Eng. J. **60**, 435-449 (2012)]. The equivalent source modeling parameters (the distributions' peak locations, amplitudes, widths, and the relative phase angle between correlated sources) are selected using Bayesian optimization implemented with simulated annealing and fast Gibbs sampler algorithms. The resulting equivalent source model reasonably predicts the radiated midfield up to 1250 Hz [Hart *et al.*, POMA **19**, 055094 (2013)]. In this study, the relationship between the correlated array's peak location and its phase angle has been further analyzed. Although sensitivity analysis of the results reveals non-uniqueness of the model, it also yields additional physical insight in the form of bounds for the dominant aeroacoustic source region as a function of frequency. The far field sound radiation predicted by the equivalent source model for a wide range of frequencies will be compared to measured far-field directivities. [Sponsored by the Office of Naval Research.]

1:20

3pPAa2. Acoustical holography and proper orthogonal decomposition analyses of full-scale jet source properties. Alan T. Wall (Dept. of Phys. and Astronomy, Brigham Young Univ., Bldg. 441, Wright-Patterson AFB, Ohio 45433, alantwall@gmail.com), Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Research and Consulting, Asheville, NC)

Efforts to characterize and reduce noise emissions from high-performance fighter aircraft are often focused on the modeling and control of large-scale turbulence structures within the jet exhaust. In past investigations, these structures have been represented as oscillatory functions whose amplitudes grow then decay with distance from the jet nozzle, or wave packets. Recently, acoustical holography methods have been used to reconstruct the radiated field of a full-scale jet, including field properties at the source. Proper orthogonal decomposition of the reconstructed source region, based on the multiple signal classification (MUSIC) algorithm, provides a physically intuitive set of partial sources. Taken together, this set is an equivalent source representation of the large-scale structures. Individually, each partial source exhibits similar qualitative behavior to that of wave packet source models.

1:40

3pPAa3. Wavepacket noise source model for microphone array data analysis of hot supersonic jets. Philip Morris (643 Berkshire Dr., State College, PA 16802, pjm@psu.edu), Robert Dougherty (OptiNav, Inc., Bellevue, WA), Chris Nelson (ITAC, LLC, Lynnwood, Washington), Alan Cain (ITAC, LLC, Chesterfield, MO), and Kenneth Brentner (State College, PA)

Phased arrays of microphones have proved themselves to be a powerful tool for aeroacoustic investigations. There are many different algorithms for processing the resulting data, including classical beamforming, its modern derivatives, Linear Programming, and Generalized Inverse Methods. The current work stems from a recognition that, for configurations with extended coherent sources (such as hot supersonic jets), the Generalized Inverse Method may be preferred, but that its accuracy can be improved by improving the underlying source model that it uses. We examine a wavepacket-based source model for analysis of the noise emitted from hot supersonic jets. This approach provides a more physically realistic representation of the jet noise sources than previously used. The model is tested using data obtained from numerical simulations as measured at a "virtual" array of microphones. The resulting generalized inverse method analysis is then used to predict noise at a farfield arc, and this prediction is compared with that from a conventional Ffowcs Williams-Hawkings acoustic analogy prediction. Initial results with the new wavepacket source model are encouraging, with improved directivity predictions and elimination of some spurious noise sources. The results of the ongoing model development will be included in the final paper.

2:00

3pPAa4. Large eddy simulation of crackle noise in hot supersonic jets. Joseph W. Nichols (Aerosp. Eng. and Mech., Univ. of Minnesota, 107 Akerman Hall, Minneapolis, MN 55455, jwn@umn.edu) and Sanjiva K. Lele (Aeronautics and Astronautics, Stanford Univ., Stanford, CA)

Crackle noise from heated, supersonic jets is investigated through high-fidelity large eddy simulation (LES). Simulations of a military-style nozzle reveal that N-shaped waves responsible for crackle noise emerge directly from the supersonic jet turbulence, and thus do not depend solely on nonlinear acoustic propagation effects. Conditional averaging using a backtracking algorithm furthermore reveals intermittent large-scale flow structures embedded in the jet turbulence as a likely source of the N-shaped waves. The skewness of pressure (Ffowcs Williams *et al*., 1975) and the skewness of the time derivative of pressure (Gee *et al*., 2007) are evaluated from full-field simulation data, and assessed as metrics of crackle. We test the hypothesis that the crackle level depends solely on the jet velocity (Ffowcs Williams *et al*., 1975) by comparing three simulations sharing the same geometry, but having different operating conditions so that the velocity and temperature are varied independently.

2:20

3pPAa5. Unstructured large eddy simulations of heated supersonic twin jets. Guillaume A. Brès, Frank E. Ham (Cascade Technologies, 2445 Faber Pl., Ste. 100, Palo Alto, CA 94303, gbres@cascadetechnologies.com), and Sanjiva K. Lele (Stanford Univ., Stanford, CA)

Unstructured large eddy simulations are performed with the compressible flow solver "Charles" developed at Cascade Technologies, for heated supersonic over-expanded jets issued from a twin nozzle. The study focuses on the modeling of the internal flow and its effects on the flow-field in the jet plume and ultimately on the radiated noise. In this work, near-wall adaptive mesh refinement, synthetic inflow turbulence and wall modeling are used inside the nozzle. In addition to the converging-diverging nozzle geometry, the computational domain includes the Y-duct, S-ducts, and angle adapter upstream of the nozzles, to realistically reproduce the elements of the experimental configuration where the internal flow conditions and wall modeling can be expected to affect the external flow-field and sound-field predictions. Comparisons between the numerical predictions and experimental PIV and far-field noise measurements from NASA Glenn Research Center will be presented and discussed.

Contributed Papers

2:40

3pPAa6. Infrasonic emissions from aircraft wake vortices: Field installation. Qamar Shams, Howard K. Knight (NASA Langley Res. Ctr., Hampton, VA), and Allan J. Zuckerwar (Analytical Services & Mater., Hampton, VA)

An infrasonic field installation was set up at Newport News-Williamsburg International Airport in early 2013. The system is made up of three PCB 377M06 microphones installed into non-porous subsurface windscreens [POMA 1pNS9, **18**, 040005 (2013)], which limit the bandwidth to 100 Hz. The microphones are placed 250 ft (76.2 m) orthogonal to the runway and 200 ft (60.96 m) apart. The data acquisition system is the B&K Pulse, from which time histories, spectra, and coherence between microphone channels are derived. The system is placed inside an instrumentation vehicle just behind the center microphone. Perforated drainage hoses are installed from the subsurface windscreens to adjacent drainage ditches and weight is added to the windscreens for additional stability. The drainage system has proved successful even on occasions of heavy downpour, revealing a truly all-weather system. A pistonphone calibration at 14 Hz in the field reveals that the three channels are matched to within 2 dB. This capability permits long-term monitoring of the health of the system. A sample time history of signals received from an aircraft takeoff will be presented.

2:55

3pPAa7. Infrasonic emissions from aircraft wake vortices: Experimental results. Allan J. Zuckerwar (Analytical Services & Materials Inc., 107 Res. Dr., Hampton, VA 23665, ajzuckerwr@yahoo.com), Qamar Shams, and Howard K. Knight (NASA Langley Res. Ctr., Hampton, VA)

Infrasonic emissions from aircraft wake vortices were investigated at the Newport News-Williamsburg International Airport early in the year 2013. Signals received by the microphones situated along an airport runway were processed in 10-s intervals. As an aircraft accelerates toward takeoff, it produces a large pressure burst as it passes each microphone. Following the burst, there appear low-frequency signals of high coherence among microphone pairs. These are interpreted as emissions from the aircraft wake vortices, as suggested by theory. In successive 10-s intervals, the coherence gradually diminishes to background levels, signifying the disappearance of the vortices. On landing the intervals of high coherence precede the bursts at aircraft touchdown, and then diminish. The pressure burst serves as a time stamp for the ensuing vortex emissions and thereby permits the tracking of successive takeoff or landing events on the same runway or on adjacent runways. The emission spectrum is essentially broadband, lacking spectral features (e.g., tones). Data were taken for takeoff of Airbus 319, DC-9, MD-88, CRJ, Lear Jet, Corporate Jet, and Dash-8 aircraft, and for landing of the Airbus 319. The pattern of pressure burst, high-coherence intervals, and diminishing-coherence intervals was observed for all takeoff and landing events without exception.

Session 3pPAb

Physical Acoustics: Topics in Geophysical Acoustics

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

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

Contributed Papers

1:00

3pPAb1. Backscattering and attenuation mechanisms in solids and solid-liquid suspensions. Paul Panetta (Applied Research Associates, Inc., Mail: P.O. Box 1346, Shipping: Rte. 1208 Greate Rd., Gloucester Point, VA 23062, ppanetta@ara.com)

The ultrasonic backscattering and attenuation are commonly used to characterize the properties of solids and solid-liquid suspension to determine grain morphology for solids and particle size and solids loading for solid liquid suspensions. An ultrasonic field is attenuated by absorption and scattering mechanisms as the field traverses a material. However, the relative strength of the absorption, single scattering and multiple scattering contributions are often unknown. In solids, the grain morphology and the dislocation properties are especially important contributions, and in solid-liquid suspension, the particle size and concentration control the attenuation. This paper will present a study of the attenuation mechanisms in solids and solid-liquid suspensions utilizing traditional attenuation, backscattering, and resonance or diffuse field measurements of the attenuation. The results provide the potential to separate the multiple scattering, single scattering, and absorption contribution to the various ultrasonic attenuation measurements on stainless steel alloys and solid liquid suspension. Results for solids and solid-liquid suspensions which elucidate the interrelationship between these energy loss mechanisms will be reported. Where appropriate, the experimental measurements will be compared with theoretical predictions.

1:15

3pPAb2. Sound speed and frequency-dependent attenuation determination in highly attenuating lubrication fluids. Blake Sturtevant and Dipen N. Sinha (Mater. Phys. and Applications, Los Alamos National Lab., MPA-11, MS D429, Los Alamos, NM 87545, bsturtev@lanl.gov)

Acoustic characterization of lubricating drilling muds is essential for the design of acoustics-based sensors and imaging devices for downhole petroleum or geothermal well environments. This work reports on the measurement of sound speed and the determination of acoustic attenuation in drilling muds with densities ranging from 10 to 15 pounds/gallon. Measurements were made in a two transducer pitch-catch configuration as a function of distance up to 40 cm and as a function of frequency up to 1 MHz. Corrections for diffraction will be discussed. Experimentally determined data are compared with previously reported attenuation values at selected frequencies, specifically 180 kHz and 280 kHz, and found to be in good agreement. The dB/cm attenuation values for the muds studied in this work were found to be three and five orders of magnitude greater than those values for silicone oil and water, respectively, at the same frequencies

1:30

3pPAb3. Design and implementation of a passive acoustic bedload monitoring surrogate system. Bradley Goodwiller, James Chambers (Mech. Eng., The Univ. of MS, 1 Coliseum Dr., Rm. 1079, University, MS 38677, btgoodwi@olemiss.edu), Wayne Carpenter (National Ctr. for Physical Acoust., The Univ. of MS, University, MS), Daniel Wren, Roger Kuhnle, Jr. Rigby (National Sedimentation Lab., Oxford, MS), and Robert Hilldale (Tech. Service Ctr., U.S. Bureau of Reclamation, Denver, CO)

Various methods of employing passive acoustics to monitor bedload transport by listening to noise generated by colliding gravel have been explored in both the lab and field. Expanding upon this research, a hydrophone-based passive bedload-monitoring system was designed, tested, and deployed by researchers at the University of Mississippi and the National Sedimentation Laboratory in Oxford, MS. A series of laboratory experiments was used to measure the dependence of acoustic propagation on various parameters such as depth of the water and placement of the hydrophones. These tests employed a calibrated acoustic transmitter to broadcast tones of various frequencies across a gravel bed. Additional flume tests used a motorized carrier to drag rocks across the gravel bed, allowing a known sediment flux to be measured acoustically without any flow noise. The hydrophone-based system was deployed on the Trinity River, Weaverville, CA, in the summer of 2012 and the Elwha River, Port Angeles, WA, in the summer of 2013. Both deployments were co-incident with physical bedload measurements taken by Graham Mathews and Associates. The design of the acoustic system, methods used to analyze the data, results from laboratory experiments, and preliminary results from field data collection will be presented.

1:45

3pPAb4. Measurement and modeling of pulsed bi-frequency, nonlinear acoustic excitation of buried landmines. Benjamin J. Copenhaver, Justin D. Gorhum, Charles M. Slack, Martin L. Barlett, Thomas G. Muir, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, bcopenhaver@utexas.edu)

To help resolve certain practical issues with acoustical methods for humanitarian landmine detection, we have researched using a pulsed, standoff source method for acoustical excitation of the buried mine [J. Acoust. Soc. Am. **130**, 2541 (2011); J. Acoust. Soc. Am. **133**, 3457 (2013)]. Pulses consisting of two primary frequencies are used in order to search for induced nonlinear vibrations at interaction frequencies such as the sum frequency, which arise due to nonlinear interaction at the mine/soil interface. To model the pulsed excitation, we employ a fully nonlinear time-domain

3p WED. PM

implementation of the lumped-element model of nonlinear soil/mine interaction introduced by Donskoy *et al.* [J. Acoust. Soc. Am. **117**, 690 (2005)]. Modeling is compared with experimental results, which are obtained with bi-frequency pulses exciting a soil with a buried landmine replica, instrumented with a geophone and a nearby microphone. Cases investigated include: (1) target only, (2) buried target under disturbed soil, (3) disturbed soil only, and (4) undisturbed soil. Excitation both on and off the resonance of the buried mine is also investigated, as is burial in different soil types at various depths. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

2:00

3pPAb5. A unified model of hysteresis and relaxation in geophysical materials. Lev A. Ostrovsky (PSD, NOAA ESRL, 325 Broadway, R/PSD99, Boulder, CO 80305, lev.a.ostrovsky@noaa.gov) and Andrey V. Lebedev (Inst. of Appl. Phys., Nizhni Novgorod, Russian Federation)

A physical model of stress-strain dynamics and long-time relaxation (slow time) in structured media such as the rock is suggested. It is based on an adhesion mechanism (JKR model) for inter-grain contacts which implies the surface force potential with a barrier. This model results in a unified description of the classical nonlinearity, stress-strain hysteresis, and logarithmic relaxation law for sound velocity and, hence,for the frequency of nonlinear resonance in samples of structured materials. Estimates of a characteristic volume of interacting contacts give close values for a variety of consolidated materials. Propagation of waves in such materials is briefly considered. For a weak (linear) testing wave, the logarithmic relaxation occurs if the classical quadratic nonlinearity is added to the stress-strain relation.

2:15

3pPAb6. A hybrid geostatistical-acoustical model for estimating single-event noise levels from noise monitor data. Edward T. Nykaza (ERDC-CERL, 2902 Newmark Dr., Champaign, IL 61822, edward.t.nykaza@usace.army.mil) and Anthony A. Atchley (Acoust., The Penn State Univ., University Park, PA)

A hybrid geostatistical-acoustical (geo-acoustic) model is proposed as a method for estimating single-event noise levels over a large region from data obtained from a small number of noise monitors. The geo-acoustic model is developed using geostatistical theory and an environmental acoustic-propagation-based regression model. The model is compared to several benchmark models and is evaluated under controlled and simulated meteorological conditions, noise monitor geometries, and areal sensor densities. The results show that it is possible to obtain accurate estimates of the SPL and the variance—associated with the SPL estimates—over a large region with a small number of noise monitors. The proposed geo-acoustic model is significantly more accurate than other commonly used spatial interpolation models, especially when there are few noise monitors and when the estimation point is extrapolated from the noise monitor data.

2:30

3pPAb7. Frequency decorrelation of broadband acoustic signals in a turbulent atmosphere. Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., 325 Broadway, Boulder, Colorado 80305, vladimir.ostashev@noaa.gov), D. Keith Wilson, and Sergey N. Vecherin (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

The impact of atmospheric turbulence on sound propagation is an important consideration for source localization with acoustic sensor arrays, studies of noise pollution, and the development of new remote sensing techniques. This paper takes as a starting point a set of recently derived, closed-form equations for the spatial-temporal coherence function of a broadband acoustic signal propagating in a refractive, turbulent atmosphere with spatial-temporal fluctuations in temperature and wind velocity. The theory is quite general and enables analysis of many statistical characteristics of the sound field. It has certain advantages in comparison with Monte-Carlo simulations and has already been used to study the spatial-temporal coherence of narrowband signals. In the present paper, this theory is employed to calculate and analyze the frequency decorrelation of broadband acoustic signals for typical regimes of the atmospheric boundary layer: mostly cloudy or sunny, with light, moderate, or strong wind. The results obtained are then used to study the effect of atmospheric turbulence on the temporal broadening of the intensity of an acoustic pulse.

2:45

3pPAb8. A combined finite element/parabolic equation formulation for acoustic wave propagation over and within a rigid porous ground. Hongdan Tao and Kai Ming Li (School of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47906, mmkmli@purdue.edu)

This paper describes a finite-element (FEM) marching scheme for a standard wide-angle parabolic equation (PE) formulation for calculating the sound fields over and within a rigid porous ground. The study has an application for exploring the effect of snow cover on the propagation of horizontal acoustic waves under refractive atmospheric conditions. Instead of using the standard Crank Nicholson method to solve for the finite difference marching scheme, a FEM discretization approach has been advocated in the present study. By using this approach, the boundary conditions, i.e., the continuity of acoustic pressure and velocity, can be incorporated directly at the air/ground and ground/ground interfaces that facilitates the simultaneous calculations of the sound fields over and within the rigid porous ground. The FEM/PE formulation yields a complete set of information on the acoustic pressure and its spatial derivatives at any receiver locations. In addition, the perfectly matched layer (PML) technique has been adopted that simulates a dissipative layer especially designed to absorb sound waves without reflections from the uppermost layers. Numerical simulations have confirmed that the application of such impedance-matched PML has efficiently eliminated reflections and reduced the overall computational time.

Session 3pSA

Structural Acoustics and Vibration, ASA Committee on Standards, and Engineering Acoustics: Structural Health Monitoring II

Tribikram Kundu, Cochair
Civil Eng. & Eng. Mech., Univ. of Arizona, 1209 E. 2nd St., Bldg. # 72, Tucson, AZ 85721

Anthony Croxford, Cochair
Univ. of Bristol, Queens Building, Bristol BS8 1TR, United Kingdom

Invited Papers

1:00

3pSA1. High frequency guided ultrasonic waves for defect monitoring. Paul Fromme (Mech. Eng., UCL, Torrington Pl., London Wc1E 7JE, United Kingdom, p.fromme@ucl.ac.uk)

High frequency guided ultrasonic waves offer a good compromise between area coverage and defect detection sensitivity for the health monitoring of critical sections of structures. Using standard ultrasonic transducers with single sided access to the structure, guided wave modes were generated that penetrate through the complete thickness of the structure. The wave propagation and interference of the guided wave modes depends on the frequency thickness product. Laboratory experiments were conducted using a laser interferometer and the wave propagation dependence on the wall thickness reduction verified against theoretical predictions. Measurements were conducted using accelerated corrosion in a salt water bath. From the measured signal changes due to the wave mode interference the wall thickness reduction due to corrosion was monitored. The energy transfer between the plate surfaces results in a good sensitivity for the detection of small defects on both surfaces. Experimentally the reflected wave was measured using standard pulse-echo equipment. Using a combination of evaluation in the time and frequency domain, the defect location and damaged plate side can be accurately determined. The high frequency guided waves have the potential for damage monitoring at critical and difficult to access locations from a stand-off distance.

1:20

3pSA2. Ultrasonic transducers and monitoring methods for high resolution structural health monitoring. Wolfgang Grill (Inst. of Experimental Phys. II, Univ. of Leipzig, Burgweg 8, Koenigstein im Taunus, Hessen 61462, Germany, wg@analogspeed.de)

An overview is presented covering novel methods for structural health monitoring (SHM) including the scientific background of different, lately developed methods, dominantly based on the transport properties of guided ultrasonic waves. Among the issues presented and discussed are mode and velocity selective contact and non contact transducers including testing methods for array transducers allowing, respectively, adapted optimized operation. Principles of operation, including holographic and tomographic imaging and high resolution integral temporal monitoring, are presented. Special attention is given to wideband excitation in combination with spectroscopic and dispersive imaging based on respective data acquisition and evaluation. Furthermore, compensation methods for temperature variations by acoustic thermometry are presented. The advantages and possible shortcomings of locally selective and integral SHM are discussed and examples are presented for aircrafts, civil structures, and their sections.

1:40

3pSA3. Guided wave structural health monitoring using inductively coupled embedded sensors. Anthony Croxford, Chenghuan Zhong, and Paul Wilcox (Mech. Eng., Univ. of Bristol, Queens Bldg., Bristol BS8 1TR, United Kingdom, a.j.croxford@bristol.ac.uk)

Conventional SHM systems typically rely on permanently attached sensor networks glued on to a structure. These add complexity and weight through either a wiring requirement or the use of wireless sensing nodes. This paper reports on an alternative approach whereby the wire between transducer and ultrasonic equipment is replaced by an inductive coupling. This allows a passive small sensor unit to be embedded into a composite component. Here a model is presented to describe the performance and optimization of such a coupling. The resulting sensors are embedded in a CFRP component and shown to exhibit excellent performance. Signal processing to account for the effects of misalignment is described. Finally, the ability of such a system to detect typical impact damage is demonstrated.

2:00

3pSA4. Inspection vs structural health monitoring: Manual ultrasonic thickness measurements compared to monitoring with permanently installed sensors. Frederic B. Cegla, Peter E. Huthwaite, and Michael J. Lowe (Mech. Eng., Imperial College London, NDE Group, Mech. Eng., London SW7 2AZ, United Kingdom, p.huthwaite@imperial.ac.uk)

Corrosion is a major issue in industry and inspection and monitoring for wall thickness loss are important to assess the structural integrity of pipework and process vessels. Manual ultrasonic thickness measurements are widely used; however, they are also notoriously unreliable because of operator errors. Therefore, automated inspection scans and monitoring at fixed locations with permanently installed sensors are becoming more attractive; they help to remove some of the error introduced by operators. However, this raises the question of what the underlying uncertainties of automated ultrasonic wall thickness measurements are. A key contributor to the uncertainty is the surface roughness condition and the authors have been researching this topic for some time. This talk will give an overview of the physics of scattering of ultrasonic waves from rough corroded surfaces. The different scales of roughness will be discussed, and a simulation technique based on the Distributed Point Source Method (DPSM) to model the scattering and its experimental validation will be presented. The need for statistical results makes both the speed and accuracy of the simulation very important. Finally, based on the simulations, results of the estimated ultrasonic measurement errors due to roughness are presented.

Contributed Paper

2:20

3pSA5. Relationship between internal damping and modal shapes due to fracture growth. Jose Villalobos (Mater. Eng., Univ. Autonomous of Nuevo Leon, Av. Universidad S/N,, Ciudad Universitaria, San Nicolas de los Garza, NUevo Leon 66451, Mexico, villalobosluna@gmail.com), Diego Ledezma (Mech. Design, Univ. Autonomous of Nuevo Leon, San Nicolas de los Garza, Nuevo Leon, Mexico), and Moises Hinojosa (Mater. Eng., Univ. Autonomous of Nuevo Leon, San Nicolas de los Garza, Nuevo Leon, Mexico)

It is shown in this work that the modal shapes and internal damping of simple structures such as bars change when a crack is present in the system, due to geometry changes. The amount of change in damping could be related to crack propagation and detected with traditional damping measurement techniques. Several amorphous materials, metals, and ceramics are subjected to a crack with controlled propagation and their internal damping is measured with modal analysis techniques. It is shown there is a potential application of modal analysis techniques for crack detection in simple geometries.

WEDNESDAY AFTERNOON, 4 DECEMBER 2013 PLAZA B, 1:00 P.M. TO 2:30 P.M.

Session 3pSC

Speech Communication: Communication Disorders

Alexander L. Francis, Chair
Purdue Univ., SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907

Contributed Papers

1:00

3pSC1. The effects of speech compression algorithms on the intelligibility of dysarthric speech. Rene L. Utianski (Dept. of Speech and Hearing Sci., Arizona State Univ., Coor Hall 2211, 10th St. and Myrtle, P.O. Box 870102, Tempe, AZ 85287-0102, rutiansk@asu.edu), Steven Sandoval, Visar Berisha (Dept. of Elec. Eng., Arizona State Univ., Tempe, AZ), and Julie Liss (Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Reduced vowel space area has been demonstrated in dysarthria, as a result of a variety of neurodegenerative diseases and subsequent decreased motor control. The effects of this reduced vowel distinctiveness on intelligibility has been studied extensively in speakers who produce less acoustically contrastive vowels, resulting from dysarthria, or even speaking casually. Past results have demonstrated that when examining such speech, listeners better identify acoustically distinctive vowel tokens. Given this, and the notion of expanded vowel space facilitating vowel identification, speech compression algorithms will be utilized to heighten the distinction between tense and lax vowels. Both connected speech and forced-choice hVd context will be tested and the effects on vowel identification and confusion will be assessed. Given the importance of vowels to overall communication, the proposed method may offer improved intelligibility when speakers cannot improve their speech production. Results bear on the possible utility of exaggerated speech compression algorithms as an augmentative communication tool for individuals with motor speech disorders.

1:15

3pSC2. Predictions from"speech banana"and audiograms: Assessment of hearing deficits in Thai hearing loss patients. Nittayapa Klangpornkun (Dept. of Elec. and Comput. Eng., Faculty of Eng., Thammasat Univ., 99, Moo 18, Phaholyothin Rd., Klongnung, Klongluang, Pathumthani 12120, Thailand, nittayapa@gmail.com), Chutamanee Onsuwan (Dept. of Linguist, Faculty of Liberal Arts, Thammasat Univ., Pathumthani, Thailand), Charturong Tantibundhit (Dept. of Elec. and Comput. Eng., Faculty of Eng., Thammasat Univ., Pathumthani, Thailand), and Pittayapon Pitathawatchai (Dept. of Otorhinolaryngology, Faculty of Medicine, Thammasat Univ., Pathumthani, Thailand)

"Speech banana" is a banana-shaped plot of speech power distribution, where the abscissa and ordinate represent frequency and intensity. By superimposing speech banana over an audiogram, tested with pure tones, degrees

of gain or loss of individual speech sound could be predicted. Speech banana has been constructed for English (Northern and Downs, 1984) and Swedish (Liden and Fant, 1954); however, none has been proposed for tonal languages, such as Thai. This work presents a construction of speech banana for Thai, a language with 21 consonants and 5 lexical tones. Specifically, intensity of each phoneme in the speech banana was calculated by differences of sound pressure level between the local maxima of power spectral density and equal loudness contour at 0 dB. Distribution of the 21 consonants is around 170-5700 Hz and 25-65 dB. Predictions of gain or loss of the phonemes from the constructed speech banana and audiograms were evaluated based on perception test results from seven Thai sensori-neural hearing loss patients, where they identified what they heard from a pair of rhyming words (210 stimuli) differing in initial phonemes, equally distributed across phonemes. Interestingly, the results showed high prediction rates of 71.4-85.7% for phonemes predominantly emphasized on frequency below 2000 Hz.

1:30

3pSC3. Effects of vowel duration and increasing dynamic spectral information on identification of center-only and edges-only syllables by cochlear-implant users and young normal-hearing listeners. Catherine L. Rogers, Gail S. Donaldson, Lindsay B. Johnson, and Soo Hee Oh (Commun. Sci. and Disord., Univ. of South Florida, USF, Dept. of Comm. Sci. & Dis., 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

In a previous study, cochlear implant (CI) users' vowel-identification performance was compared to that of young normal-hearing (YNH) listeners. Stimuli included full syllables and two duration-neutralized conditions: center-only and edges-only (silent-center). CI users performed more poorly than YNH listeners overall and showed proportionately larger decrements in performance for partial syllables. Error analyses suggested that at least some CI users rely more heavily on vowel-duration cues than YNH listeners. The present study was designed to test this hypothesis and to determine whether increasing duration of dynamic cues in the edges-only conditions would improve performance, particularly among poorer-performing CI users. Ten YNH listeners and ten adult CI users heard /dVd/ syllables recorded from three talkers. Full syllables were edited to create center-only and edges-only stimuli in which vowel duration cues were or were not preserved, plus edges-only stimuli with different durations of dynamic information. Performance of both groups improved in the duration-preserved condition for center-only, but not edges-only, stimuli. The center-only duration benefit was larger for the CI than for the YNH group. Increasing the duration of dynamic information in the silent-center stimuli improved vowel-identification performance for both groups. Individual differences among CI users and implications for listener-training programs will be discussed.

1:45

3pSC4. Communicative intent and affect in mothers' speech to hearing-impaired infants with cochlear implants. Maria V. Kondaurova, Tonya R. Bergeson (Otolaryngol. – Head & Neck Surgery, Indiana Univ. School of Medicine, 699 Riley Hospital Dr. – RR044, Indianapolis, IN 46202, mkondaur@iupui.edu), and Christine Kitamura (MARCS Lab., Univ. of Western Sydney, Penrith, NSW, Australia)

Emotional properties of infant-directed speech influence normal-hearing (NH) infants' attention to speech sounds. The current study examines communicative intent/affect in speech to hearing-impaired (HI) infants following the first year of cochlear implantation. Mothers of HI infants (HI group, ages 13.3—25.5 months), NH age-matched infants (NH-AM group, ages 13.5—25.7 months) and NH experience-matched infants (NH-EM group, ages 2.3—3.6 months) were recorded playing with their infants at three sessions over the course of one year. 25-second speech samples were low-pass filtered, leaving pitch but not speech information intact. Twelve adults rated stimuli along five scales of communicative intent/affect: Positive/Negative Affect, Intention to Express Affection, Encourage Attention, Comfort/Sooth and Direct Behavior. ANOVAs demonstrated main effects of Group and/or Session for all scales (ps = 0.01 to 0.07). Speech to HI and NH-EM infants was more positive, affective, encouraging, and comforting than speech to NH-AM infants. Mothers decreased affective (NH-EM group) and comforting (HI group) speech qualities over three sessions but increased directive behavior (NH-EM group). The results suggest that affective properties are modified in speech to HI infants depending on their hearing experience rather than chronological age. Mothers adjust these properties to their infant's developmental stage over the 12-month period.

2:00

3pSC5. Feature divergence of pathological speech. Steven Sandoval (School of ECEE, SenSIP Ctr., Arizona State Univ., 2323 E Apache Blvd., Apt. 2120, Tempe, AZ 85281, ssandova@gmail.com), Rene Utianski, Visar Berisha, Julie Liss (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ), and Andreas Spanias (School of ECEE, SenSIP Ctr., Arizona State Univ., Tempe, AZ)

Many state of the art speaker verification systems are implemented by modeling the probability distribution of a feature set using Gaussian mixture models. In these systems, a decision is made by comparing a likelihood of an observation using both a Gaussian mixture model corresponding to an individual, and a Gaussian mixture model universal back ground model. In this study we propose to use a similar framework to instead characterize the divergence of the feature set distribution between healthy and pathological speech. We accomplish this by determining the difference between a universal background model trained on healthy speech and model of an individual's pathological speech. There are several known methods to evaluate the difference between two probability distributions, one example being the Kullback-Leibler divergence. By building a universal background model using healthy speech, we hope to capture the expected distribution of our feature space. Then by computing a difference between a dysathric individual's feature distribution, and the universal background model, we can determine the features that are most likely to capture the effects of a specific motor speech disorder.

2:15

3pSC6. The functional impact of incidental orofacial muscle activity. Lauren R. Johnson, Nancy L. Potter, and Mark VanDam (Speech & Hearing Sci., Washington State Univ., Spokane, WA)

Repetitive use of specific muscle groups is known to increase both strength and the ability to sustain muscle activity (i.e., endurance) of those muscle groups. Certain orofacial muscles are necessarily recruited in the course of playing a brass instrument, and thus regular performers may incidentally gain strength and endurance in the orofacial muscles used to perform. To test this possibility, 16 skilled trumpet players and 16 non-playing controls contributed strength and endurance (at 50% of maximum strength) measures for buccal, lingual, and labial muscle groups. Results indicate that trumpet players had greater cheek strength and lip endurance, but there were no differences between test and control groups for tongue strength or endurance. Findings suggest that incidental orofacial muscle activity may have a positive functional impact on orofacial muscle strength and/or endurance. This finding supports a clinically useful, objective measure for diagnosis and may be useful for functional rehabilitation for patients with orofacial disorders including those with Bell's palsy, complications associated with otitis media, acoustic neuromas, or other facial- or cranial nerve damage due to surgery, trauma, or disease.

Session 3pSP

Signal Processing in Acoustics: Smartphone Acoustic Signal Processing Student Competition (Poster Session)

Kevin Cockrell, Chair
Applied Physical Sciences Corp., 2488 Historic Decatur Rd., Ste. 100, San Diego, CA 92106

Contributed Papers

All posters will be on display and all authors will be at their posters from 1.30 p.m. to 2.30 p.m.

3pSP1. SoundMap: A mobile app for optimizing room acoustics and speaker placement. K. J. Bodon and Zachary Jensen (Phys., Brigham Young Univ., Brigham Young Univ., Provo, UT 84602, joshuabodon@gmail.com)

Improving room acoustics can be expensive and time consuming, even for the most experienced acousticians. SoundMap ™ is a user friendly app that can be used to easily model rooms based upon acoustic properties. A combination of the method of images, ray tracing, and modal analysis will be used to calculate steady state levels for different frequency bands in a room. The modeling will also take several other factors into account, including the room impulse response, absorption coefficients of boundary materials, furniture, and other room elements. The model will incorporate generic loudspeaker systems, with the future design goal of integrating specific loudspeaker systems. Through a contour plot, the user will be able to test the effects of loudspeaker location, loudspeaker type, room layout, and room materials, with the aim of optimizing the listening experience.

3pSP2. Ping-pong: Using smartphones to measure distances and relative positions. Jorge Herrera and Hyung-Suk Kim (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 660 Lomita Ct., Stanford, CA 94305, jorgeh@ccrma.stanford.edu)

A novel system for real-time range and geometry estimation of a group of smartphones co-located in a shared physical space is presented. The system uses off-the-shelf devices and employs audible signals to estimate inter-device (pair-wise) distances. Coordinated sound synthesis and processing of a pair of pitched sounds allows to estimate the distance between two devices based on the travel time. By independently analyzing different harmonics of the sounds used in the measurement, a more robust and precise distance estimation is achieved. To overcome the absence of a centralized clock to coordinate measurements, a synchronous communication channel was used. When four or more devices are present, it is possible to estimate their relative positions in a three-dimensional space, by minimizing an equation error norm. The system works both on closed and open spaces. We believe that such system opens the possibility for new ways of interaction that could benefit musical expression, social interaction and gaming.

3pSP3. Speech assist: An augmentative tool for practice in speech-language pathology. Rene L. Utianski (Dept. of Speech and Hearing Sci., Arizona State Univ., Coor Hall 2211, 10th St. and Myrtle, P.O. Box 870102, Tempe, AZ 85287-0102, rutiansk@asu.edu), Steven Sandoval (Dept. of Elec. Eng., Arizona State Univ., Tempe, AZ), Nicole Lehrer (Dept. of Arts, Media, and Eng., Arizona State Univ., Tempe, AZ), Visar Berisha (Dept. of Elec. Eng., Arizona State Univ., Tempe, AZ), and Julie Liss (Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Dysarthria affects approximately 46 million people worldwide, with three million individuals residing in the US. Clinical intervention by speech-language pathologists (SLPs) in the United States is supplemented by high quality research, clinical expertise, and state of the art technology, supporting the overarching goal of improved communication. Unfortunately, many individuals do not have access to such care, leaving them with a persisting inability to communicate. Telemedicine, along with the growing use of mobile devices to augment clinical practice, provides the impetus for the development of remote, mobile applications to augment the work of SLPs. The proposed application will record speech samples and provide a variety of derived calculations, novel and traditional, to assess the integrity of speech production, including: vowel space area, assessment of an individual's pathology fingerprint, and identification of which parameters of the intelligibility disorder are most disrupted (e.g., prosody, vocal quality). The individualized selection of desired information for incorporation into a report template will be available. The reports will mimic those generated manually by SLPs today. The automation of this assessment will allow SLPs to treat patients remotely, allowing for the widespread, worldwide impact of high skilled assessment, something currently lacking in underdeveloped parts of the world.

3pSP4. SpeakerLab: A mobile app for measuring loudspeaker parameters and modeling enclosures. Zachary Jensen and K. J. Bodon (Brigham Young Univ., 934 E 300 S, Provo, UT 84606, zjens1@gmail.com)

Thiele-Small parameters for loudspeaker drivers are essential in modern loudspeaker design. While they are typically given in specifications by a driver manufacturer, their values can vary considerably. SpeakerLab is a loudspeaker parameter measurement and modeling tool, an all-inclusive app for the loudspeaker designer. Using a specially designed cable with a known impedance, a user will be able to plug their mobile device directly into a loudspeaker driver and measure its Thiele-Small parameters using the added mass or known volume method. With the correctly measured parameters, an enclosure can be appropriately modeled and optimized.

3pSP5. "Tone deaf"— The touch based ear-training game. Andrew T. Pyzdek (Appl. Res. Lab., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu)

Inspired by the color matching game "Colorblind" by ChewSoft, Tone Deaf is the new game that challenges players to match the pitch and timbre of a note by touching the screen to draw a single cycle of the waveform. The shape of the waveform is then used to determine the appropriate harmonics and their phases, while the length of the cycle determines the frequency. Players will progress through multiple levels of difficulty, starting with simple waveforms such as sine waves, sawtooth waves, and square waves, and finally emulating the complex patterns of instruments such as the piano, trumpet, and violin.

3pSP6. The echolocating phone app. David A. Hague and Saurav Tuladhar (Elec. and Comput. Eng., Univ. of Massachusetts, Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, david.a.hague@gmail.com)

Acoustic echolocation systems have a multitude of applications including test/measurement, noise cancellation, and object detection. Current smartphone technology possesses the necessary computing power to implement sophisticated echolocation systems on smartphones. This provides the smartphone user

with the ability to perform complicated acoustic measurement tasks in any situation and environment. This project implements an active sonar system on an Android smartphone. The sonar system has three main modes of operation: object detection and ranging, object velocity estimation, and joint object range and velocity estimation. Each of these operational modes are accomplished by utilizing one of the phone's speakers as a transmitter and one or more of the microphones as a receiver. The app employs a simple Graphical User Interface that allows the user to switch between the modes of operation and observe/analyze object data. Additional functionality may include the ability to write object data to various data formats for offline processing and analysis. Because this app employs several modes of operations, there are several potential commercial applications including measuring a room's acoustic impulse response or a vehicle speed gun as well as many educational apps exploring various acoustic phenomena.

WEDNESDAY AFTERNOON, 4 DECEMBER 2013 CONTINENTAL 4, 3:30 P.M. TO 5:30 P.M.

Plenary Session, Annual Meeting, and Awards Ceremony

James H. Miller, President
Acoustical Society of America

Annual Meeting of the Acoustical Society of America

Presentation of Certificates to New Fellows

Judit Angster – for contributions to the acoustics of the pipe organ

Benjamin A. Cray – for contributions to underwater directional sensing

Kevin D. Heaney – for contributions to ocean acoustic modeling and inversion methods

Marcia J. Isakson – for contributions to modeling shallow water acoustic propagation using the finite element method

Tribikram Kundu – for contributions in nondestructive testing and evaluation

Robert M. Koch – for contributions to the hydroacoustics and structural acoustics of underwater systems

Michael V. Scanlon – for contributions to high speech imaging of fine scale acoustic phenomena

Clark S. Penrod – for service to the Society and leadership in underwater acoustics

Gopu R. Potty – for contributions to ocean acoustic inversion methods in shallow water

Kausik Sarkar – for contributions to the modeling of ultrasound microbubbles

Natalia A. Sidorovskaia – for contributions in research and education in underwater acoustics and animal bioacoustics

U. Peter Svensson – for contributions to room acoustics edge diffraction modeling

Jing Tian – for leadership in promoting American-Chinese cooperation in acoustics

Award Announcements and Presentations of Awards

Announcement of the 2013 Munk Award to W. Steven Holbrook
granted jointly by The Oceanography Society, the Office of Naval Research, and
the Office of the Oceanographer of the Navy

ASA Student Council Mentoring Award to Barbara G. Shinn-Cunningham

Rossing Prize in Acoustics Education to Juliette W. Ioup

Silver Medal in Biomedical Acoustics to Kullervo H. Hynynen

Silver Medal in Musical Acoustics to William J. Strong

WEDNESDAY AFTERNOON, 4 DECEMBER 2013 CONTINENTAL 5, 5:30 P.M. TO 7:30 P.M.

Session 3eED

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

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

Marcia J. Isakson, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713

This workshop for San Francisco area Girl Scouts (ages 12–17) consists of a hands-on tutorial, interactive demonstrations, and panel discussion about careers in acoustics. The primary goals of this workshop are to expose the girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please email Traci Neilsen (tbn@byu.edu) if you have time to help with either guiding the girls to the event and helping them get started (5:00 p.m. – 6:00 p.m.) or exploring principles and applications of acoustics with small groups of girls (5:00 p.m. – 7:30 p.m.). We will provide many demonstrations, but feel free to contact us if you would like to bring your own.

WEDNESDAY, 4 DECEMBER 2012 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday and Thursday evenings beginning at 8:00 p.m. and on Wednesday evening beginning at 7:30 p.m.

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

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Golden Gate 2/3
Signal Processing in Acoustics	Continental 1