Session 1aAA

Architectural Acoustics and Education in Acoustics: Computer Modeling in Buildings and the Environment as an Education Tool

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

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

Chair's Introduction—9:00

Invited Papers

9:05

1aAA1. Using computer building modeling and auralization as teaching tools. Robert C. Coffeen (School of Architecture, Design & Planning, Universtiy of Kansas, 1465 Jayhawk Blvd, Lawrence, KS 66045, coffeen@ku.edu)

Acoustic building modeling in computer programs is very useful in the understanding of room acoustics for venues of various types by architecture and architectural engineering students. Models provide calculation of reverberation time using the Sabine and similar equations as interior materials are changed. Ray tracing can be used to understand the effect of disturbing sound reflections from interior surface shapes and locations. Being able to create impulse responses in a model allows the estimation of reverberation time using Schroeder integration. And, transferring impulse responses to a measurement and analysis program allows determination of early decay time as well as T10, T20, T30 and other sound decay cutoff times. In addition, more advanced students can determine Sound Transmission Class STI, Strength G, Inter-aural Cross Correlation Coefficient IACC, and other acoustic parameters. But, one of the most useful items that can be produced by model impulse responses is auralization. This allows students to hear a simulation of room sound as reverberation time and other acoustic parameters are changed. Examples of using one of the several modeling and analysis programs will be presented.

9:25

1aAA2. Simple interactive virtual auralizations as educational tools. Christopher L. Barnobi (Stewart Acoustical Consultants, 7330 Chapel Hill Rd, Suite 101, Raleigh, NC 27607, chris@sacnc.com)

This presentation provides an overview and demonstration of some 'classic' acoustic phenomena using a computer simulation. The computer program provides a visual rendering of an environment with a source and receiver. By allowing the user to vary some of the parameters of the environment, a user can see and hear the differences in spaces by changing the surroundings. The goal is to highlight the well known environmental factors that impact sound such as volume in a room. A variety of parameters and environments will be explored.

9:45

1aAA3. Education technology in architectural acoustics: A hands-on program for teaching. Norman H. Philipp (School of Architecture, Design & Planning, University of Kansas, Lawrence, KS 66045, philipp.norman@gmail.com)

Through the implementation of educational technology a web-based educational tool is being developed to aide in the teaching of architectural acoustics to architecture students at the undergraduate and graduate level. As the first step, its scope has been limited to reverberation time in architectural acoustics. The overall objective is to provide a dynamic educational tool for both educators and students to improve their understanding and retention of the principles of architectural acoustics.

10:05–10:25 Break

10:25

1aAA4. Finite difference simulation methods as an educational tool. Jonathan Botts, Ning Xiang, and Todd Brooks (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th St., Greene Building, Troy, NY 12180, botts.jonathan@gmail.com)

Finite difference methods can be a valuable and unexpected tool in acoustics education. As a wave acoustic simulation method, it provides instructive time-domain visualizations particularly useful for illustrating broadband wave effects like diffraction and interference. Furthermore, the knowledge required to implement these simulations can be taught in just a few hours of instruction with or without calculus in contrast to several other wave acoustic methods. The exposition provides opportunities for discussion of basic numerical methods as well as the physics of wave and diffusion processes. We present example projects from students of mixed science, engineering, and music backgrounds. After only four hours of instruction, they were able to independently simulate various room geometries

with impedance boundary conditions along with a variety of other acoustical systems. Beyond strictly educational value, this is a flexible and free tool that the modern acoustician can use for research, physics-based simulation, or creation of broadband virtual sound fields.

10:45

1aAA5. Accuracy of acoustic simulations and the effects of material databases. Ronald Sauro (NWAA Labs, Inc, 90 Tower blvd, Elma, WA 98541, audio_ron@msn.com)

A discussion of the effects of material databases on the accuracy of predictions emanating from acoustic simulation programs. This brings to light the lack of inherent accuracy in these programs because of the lack of accuracy in the measurement of absorption, scattering and diffusion parameters. Some of the predictions can be shown to be off as much as 300 to 500 per cent. We look at possible corrections in these measurements and how they can improve these predictions.

Contributed Papers

11:05

1aAA6. Parallelized finite difference time domain room acoustic simulation. Cameron Fackler, Jonathan Botts, and Ning Xiang (Graduate Program in Architectural Acoustics, School of Architecture, Rensselaer Polytechnic Institute, 110 8th St, Greene Building, Troy, NY 12180, facklc@rpi.edu)

A parallelized room acoustics simulator based on the finite difference time domain (FDTD) method is developed, utilizing a Blue Gene/L supercomputer. Wave-based methods such as FDTD are desirable for use in room acoustics simulations since they account for effects such as diffraction and interference. However, such methods require large amounts of computational power and memory, especially when simulating large volumes or high frequencies. To utilize the power of modern computing systems and move toward large-scale simulations of realistic concert halls, a parallel FDTD implementation is written in the C++ programming language with the Message Passing Interface (MPI) library. The volume to be simulated is partitioned into blocks, which efficiently update shared interfaces between nearest neighbors using the Blue Gene architecture's point-to-point communication network. Several compact explicit FDTD schemes are compared using simulations of various spatial volumes, executed on varying numbers of processors. The use of a Blue Gene/L supercomputer demonstrates substantial speedup over an equivalent serial implementation.

11:20

1aAA7. Acoustic simulation of the church of San Francesco della Vigna. Braxton B. Boren (Music, New York University, New York, NY 10012, bbb259@nyu.edu) and Malcolm Longair (Cavendish Laboratory, University of Cambridge, Cambridge, Cambridgeshire, United Kingdom)

San Francesco della Vigna is the oldest church in Venice for which there is evidence that acoustic considerations were taken into account in the architectural design. Francesco Zorzi, a humanist scholar, recommended that the church have a flat wooden coffered ceiling to improve the intelligibility of the sermons preached there. But instead of Zorzi's recommended flat ceiling, the church was built with a plaster vault ceiling. Using measured acoustic data from the CAMERA project, a virtual model of the church was constructed in Odeon whose simulated parameters matched the measured values at different source-receiver combinations. After obtaining a good match to the measured values, this virtual model was then altered to reconstruct the flat ceiling recommended by Zorzi. This ceiling was then placed at the two different heights at which it might have been built in Zorzi's time. The simulations show that the more absorptive ceiling might have slightly reduced the long reverberation time in the church. However, the ceiling would have been too high to make any significant change in the D50, which still remains extremely low. Thus this simulation indicates that Zorzi's ceiling would not have made the impact on speech intelligibility he had expected.

Session 1aAO

Acoustical Oceanography and Underwater Acoustics: Memorial Session in Honor of Clarence S. Clay I

Dezhang Chu, Cochair
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John K. Horne, Cochair
School of Aquatic and Fishery Sciences, University of Washington, Seattle, WA 98195

J. Michael Jech, Cochair
Northeast Fisheries Science Center, Woods Hole, MA 02543

Timothy K. Stanton, Cochair
Woods Hole Oceanographic Institution, Woods Hole, MA 02543-1053

Chair's Introduction—8:55

Invited Papers

9:00

1aAO1. C.S.Clay: A distinguished acoustician. Ivan Tolstoy (Knocktower, Knockvennie, Castle Douglas DG7 3PA, United Kingdom, tstanton@whoi.edu)

Clay's well-known work in ocean acoustics earned him an international reputation. His contributions ranged from studies of sound propagation in shallow water to the application of filter theory in noisy environments, diffraction from rough surfaces, sound scatter by fish and, generally speaking, the execution and design of numerous experiments. Less well known, perhaps on account of his modesty, was his role at Columbia University's Hudson labs in designing a fully digitized microphone array for the study of low frequency atmospheric waves (in the 1 to 600 sec. period band), which led to the detection of gravity and acoustic-gravity waves from several nuclear tests. After leaving Columbia, Clay accepted a professorship at the University of Wisconsin where, among other things, he taught geophysics and continued research on sound scatter. His presence and participation at Acoustical Society meetings will be sorely missed.

9:20

1aAO2. C.S. Clay—A scientist of outstanding vision, brilliance, and versatility. Christopher Feuillade (Facultad de Física, Pontificia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Santiago, Región Metropolitana, Chile, chris.feuillade@gmail.com)

During a scientific career of more that 60 years, Clarence Clay made many important research contributions spanning a wide range of topics in ocean acoustics, acoustical oceanography, geophysical exploration, signal processing, SONAR system applications and techniques, and more. His achievements are detailed in over 135 peer-reviewed articles, abstracts, technical reports, and at least five patents. In addition he was the author, or co-author, of four widely read textbooks. His research work is recognized internationally for the far-reaching significance of many of the advances he made. In order to attempt a proper appreciation of the range and versatility of Clay's accomplishments, within the constraints of a 30 minute presentation, this talk will consist of a series of brief overviews of a representative selection of topics on which he worked. These include: the theory and measurement of rough surface scattering, particularly at the ocean boundaries; matched-filter signal transmission and detection, time reversal, and matched-field processing; acoustic methods and models for scattering from individual fish, and for fish detection and abundance estimation; the theory and experimental investigation of time domain scattering from wedges, and its incorporation into wedge assemblage models; and the theory and applications of seismic exploration and profiling.

9:50

1aAO3. A review of Clarence Clay's research contributions in the area of fisheries acoustics. John Ehrenberg (Hydroacoustic Technology Inc, 715 NE Northlake Way, Seattle, WA 98105, jehrenberg@htisonar.com)

This presentation provides an overview of the significant contribution that Clay and his students made to the understanding of the statistical nature of the acoustic signals scattered from fish and the methods for removing the effect of the beam pattern from the received echo statistics to measure the underlying fish backscattering statistics. By making measurements of the acoustic scattering from live fish, he showed that the probability density function of the envelope amplitude of the echo signal scattered from the fish could be modeled by a Ricean PDF. He further showed that as the ratio of the length of the fish to the acoustic wavelength became large, the PDF became Rayleigh distributed. Clay and his students were interested in using the measured echo statistics to obtain information about the size distribution of the fish producing the scattering. They developed a method for deconvolving the effect of the acoustic beam pattern from the received echo statistics to provide an estimate of the fish scattering PDF. The effectiveness of the technique was demonstrated for a fish population in a Wisconsin lake.

10:10–10:25 Break

10:25

1aAO4. Estimating numerical density of scatterers in monotype aggregations using the statistics of broadband echoes: Applications to fish echoes. Wu-Jung Lee, Timothy K. Stanton, and Andone C. Lavery (Department of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543, wjlee@whoi.edu)

The statistics of echoes from active sonar systems can yield important information on aggregations of scatterers. This study explores the use of echo statistics for estimating the numerical density of scatterers in monotype aggregations. Here, "monotype" refers to scatterers with the same scattering amplitude distribution in the considered frequency range. The signals are broadband, and the geometry involves direct paths between the sonar and the scatterers without interference from boundaries. Model probability density functions (pdf's) of envelope amplitudes of matched-filter outputs are numerically generated by varying the number of Rayleigh scatterers randomly-distributed in a half-space shell while accounting for the frequency-dependent system response, scatterer response, and beampattern effects. The shape of the echo pdf as observed by the sonar receiver is highly non-Rayleigh when there are few scatterers in the beam, and gradually approaches the Rayleigh distribution when the number of scatterers increases. This model is applied to broadband fish echoes (30-70 kHz) collected in the ocean through a best-fit procedure. The inferred numerical density of fish is comparable to the density estimated using corresponding measurements of volume backscattering strength and modeled target strengths.

10:40

1aAO5. Ice cream and the application of backscatter models. John K. Horne (School of Aquatic and Fishery Sciences, University of Washington, Box 355020, Seattle, WA 98195, jhorne@u.washington.edu)

I was invited to visit Clay and colleagues at the Center for Limnology at the University of Wisconsin, Madison in October 1991. As an acoustics neophyte, I had lots of questions that Clay patiently took the time to answer while we ate ice cream at the Memorial Union. That discussion led to the development of the Kirchhoff Ray-mode (KRM) model and increased the use of acoustic scattering models to investigate how fish reflect sound. Acoustic scattering models enable investigation of factors or conditions that cannot be replicated or isolated in field or experimental measures. The iterative combination of models with measures improves accuracy of model predictions and the understanding of how the physics of sound interacts with biology to produce acoustic data. Both the structure and application of backscatter models have evolved in their complexity and realism. Examples will be used to illustrate advances in and insight gained through modeling, with special consideration of Clay's contributions. The talk will conclude with speculation on what Clay would see as the next step.

10:55

1aAO6. Low frequency acoustical scattering properties of large schools of swim bladder fish. María P. Raveau (Facultad de Ingeniería, Pontificia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Región Metropolitana, Chile) and Christopher Feuillade (Facultad de Física, Pontificia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Santiago, Chile, chris.feuillade@gmail.com)

The collective back scattering behavior of fish schools has previously been described by a school scattering model [J. Acoust. Soc. Am., **99**(1), 196-208 (1996)], which incorporates both multiple scattering effects between neighboring fish, and coherent interactions of their individual scattered fields. In the present work, the school scattering model has been extended, and used to investigate the back- and forward-scattering properties of the acoustic field, and transmission through, large schools of swim bladder fish, at frequencies close to the swim bladder resonance frequency. Results show that their frequency and spatially-varying scattering behavior depends strongly upon the number of fish in the school ensemble, the species specific swim bladder size, the average spacing between fish, and the size and shape of the school. Results will also be presented of a comparison between the school model and fish absorption data obtained during the experiment Modal Lion, performed in the Gulf of Lion in September 1995, and reported by Diachok [J. Acoust. Soc. Am., **105**(4), 2107-2128 (1999)]. [Work supported by ONR.]

11:10

1aAO7. Acoustic characterization of thecosome pteropods and recent field measurements in the context of ocean acidification. Andone C. Lavery (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02543, alavery@whoi.edu), Gareth L. Lawson, Peter H. Wiebe (Biology, Woods Hole Oceanographic Institution, Woods Hole, MA), Timothy K. Stanton, Jonathan R. Fincke (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA), and Nancy J. Copley (Biology, Woods Hole Oceanographic Institution, Woods Hole, MA)

One of Clay's passions was modeling the scattering physics of marine organisms, a passion that has transcended into new generations of scientists. The focus of this presentation is thecosome pteropods, a widely and patchily distributed group of shelled zooplankton that are important members of pelagic ecosystems as they constitute important prey for a variety of other zooplankton and top predators. Acoustic techniques are well suited to sampling pteropods on relevant spatial and temporal scales as they secrete aragonite shells that make them highly efficient scatterers of sound. However, pteropod shells are complex and very susceptible to an increasingly corrosive seawater environment due to ocean acidification. Understanding the scattering physics is key to using acoustics as a quantitative remote sensing tool. Here we report on recent field measurements that combine the use of broadband (30-600 kHz) and narrowband (43, 120, 200, and 420 kHz) acoustic scattering techniques, as well as supporting in situ measurements (nets, optics, CTD and ocean chemistry) to investigate the distribution, abundance and size of pteropods in both the northwest Atlantic and the northeast Pacific in relation to the oceanic chemistry. Existing scattering models are tested, and improvements and modifications to the acoustic instrumentation and models are suggested.

11:25

1aAO8. Echo statistics: Pursuing one of Clay's visions. Timothy K. Stanton (Dept. of Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, MS #11, Woods Hole, MA 02543-1053, tstanton@whoi.edu) and Dezhang Chu (National Marine Fisheries Service - Northeast, National Oceanic and Atmospheric Administration, Seattle, WA)

One of the first papers that Clay gave to one of the authors (TKS) when he entered Clay's office in 1980 was one of Clay's papers involving echo statistics and, specifically, accounting for beampattern effects in using single beam echoes from resolved fish to estimate their target strength and abundance. That, and a multitude of conversations with Clay helped propel TKS, and later DC, into a career where echo statistics was an integral aspect of their work. We will review our work on echo statistics associated with a variety of scatterers–the seafloor, sea ice, fish, zooplankton, and machined objects. A key aspect of this work has been connecting the physics of the scattering process and sonar parameters with parameters of the statistical functions (such as shape parameter).

11:40–12:00 Panel Discussion

Session 1aEA

Engineering Acoustics, Signal Processing in Acoustics, and Animal Bioacoustics: Broadband, Complex Pulses for Echolocation

Kenneth M. Walsh, Chair
K+M Engineering Ltd, 51 Bayberry Ln., Middletown, RI 02842

Chair's Introduction—8:25

Invited Papers

8:30

1aEA1. Broadband synthetic aperture chirp reflection profiling. Steven Schock (Ocean and Mechanical Engineering, Florida Atlantic University, 777 Glades Road, Boca Raton, FL 33431, sschock@fau.edu), Jason Sara (Edgetech, Boca Raton, FL), and Kenneth M. Walsh (K M Engineering Ltd., Middletown, RI)

A newly developed towed chirp subbottom profiler transmits FM pulses with a bandwidth of three octaves to generate high resolution reflection profiles of the seabed. The broad bandwidth of the pulses, generated with two arrays of piston sources, produces the temporal resolution needed for resolving fine sediment layering. A 40 channel horizontal hydrophone array, embedded in vehicle wings, provides the acoustic aperture for enhancing the across track resolution of subsurface features and reducing sediment scattering noise. After application of a matched filter, synthetic aperture processing of hydrophone data generates a large aperture along the track of the sonar vehicle thereby improving along track image resolution and obtaining a further reduction in scattering noise. The reductions in backscattering from sediments yield imagery with improved subsurface penetration. The reflection profiles are stacked envelopes of coherently summed data formed by time domain focusing on planar surfaces oriented over a range of discrete slopes. This work was funded by the National Science Foundation.

8:55

1aEA2. Odontocete biosonar signals: Functional anatomy or signal design. Whitlow W. Au (HI Institute of Marine Biology, University of Hawaii, 46-007 Lilipuna Road, Kaneohe, HI 96744, wau@hawaii.edu)

There are between 67 and 76 species of odontocetes (toothed whales) and presumably all have biosonar capabilites. There are three fundamental biosonar signal types that can be categorized by types of marine mammals that produce these signals. Whales and dolphins that can emit whistle signals (except for sperm whales) project short broadband clicks containing about 5 to 7 cycles with decaying exponential envelope and Q (center frequency over bandwidth) between 2 and 3. Porpoises do not whistle and produce polycyclic narrow band high frequency biosonar signals with approximately 20 or more cycles with a modified sinusoidal amplitude envelope and Q around 14. Biosonar waveforms of beaked whales (also non-whistling animals) typically have 10-15 cycles with a linear FM component and Q around 4. This presentation will discuss the characteristics of the three different biosonar signal types and suggest some motivation factors involved with the use of the different signals. The type of prey and their habitat will also be included in the discussion. It will be shown that in some cases, the signal type is motivated by anatomical constraints of the odontocete and in other cases, the backscatter characteristics of the prey may be the most important factor.

9:20

1aEA3. Implications of the variety of bat echolocation sounds for understanding biosonar processing. James A. Simmons (Neuroscience, Brown University, 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu), Matthias Hoffmann-Kuhnt, Tzi Ming Leong (National University of Singapore, Singapore), Shizuko Hiryu, Hiroshi Riquimaroux (Doshisha University, Kyotanabe, Kyoto, Japan), Jeffrey M. Knowles (Neuroscience, Brown University, Providence, RI), and Cynthia F. Moss (University of Maryland, College Park, MD)

The variety of echolocation sounds used by different species of bats have implications for target ranging. Signals recorded at individual sites reveal species stacked in different frequency bands, perhaps to avoid cross-interference. Search-stage signals include short single-harmonic or multi-harmonic tone-bursts, or very shallow FM bursts. These narrowband sounds have abrupt onsets to evoke phasic on-responses that register echo delay, but with limited acuity. Wider FM sweeps used for searching by other bats evoke on-responses at many more frequencies for better delay acuity. These sound types may signify foraging in the open, within broad spaces bounded relatively remotely by trees or the ground. Intervals between broadcasts are consistent with biosonar operating ranges set by the boundaries of the scene in relation to atmospheric attenuation. Most species make transitions to wider signal bandwidth during interception by increasing FM sweep-width or adding harmonics. Additionally, wideband, multi-harmonic FM sounds are used by species that frequently fly vegetation; they use harmonic processing to suppress surrounding clutter and perceive the path to the front. These observations suggest basic echo-delay processing to determine target range, with increasing bandwidth first to improve delay acuity and then to determine target shape. [Work supported by ONR and NSF.]

9:45

1aEA4. Dolphins use "packets" of broadband clicks during long range echolocation tasks. James J. Finneran (US Navy Marine Mammal Program, SSC Pacific Code 71510, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil)

When echolocating, dolphins typically emit a single short duration, high-frequency, broadband "click," then wait for the echo to return before emitting another click. However, previous studies have shown that dolphins and belugas performing long-range echolocation tasks may instead emit a burst, or "packet," of several clicks, then wait for the packet of echoes to return before emitting another packet of clicks. The exact reasons for the use of packets, rather than individual clicks, is unknown. In this study, the use of packets by dolphins was examined by having trained bottlenose dolphins perform long-range echolocation tasks. The tasks featured the use of "phantom" echoes produced by capturing the dolphin's outgoing echolocation clicks, convolving the clicks with the impulse response of a physical target to create an echo waveform, then broadcasting the delayed, scaled echo waveform back to the dolphin. Dolphins were trained to report the presence of phantom echoes or a change in phantom echoes. At ranges below 75 m, the dolphins rarely used packets of clicks. For ranges greater than 75 m, the likelihood of packet use was related to both target range and echo strength. [Work supported by the SSC Pacific Naval Innovative Science and Engineering (NISE) program.]

10:10–10:25 Break

10:25

1aEA5. Environmentally neutral complex broadband biomimetic waveforms for active sonar. Peter F. Dobbins (Advanced Systems, Ultra Electronics Sonar Systems, Waverley House, Hampshire Road, Weymouth, Dorset DT4 9XD, United Kingdom, peter.dobbins@ultra-sonar.com)

There is an expanding requirement to reduce the impact of man-made sound, including active sonar transmissions, on marine mammals in the defence, offshore and other sectors. This is driven partly by increased public interest in these animals, but mainly by legislation such as the US Marine Mammal Protection Act, and similar regulatory and licensing requirements throughout the world. Typically, such requirements are met using monitoring by Marine Mammal Observers or Passive Acoustic Monitoring. Having detected animals within a specified range, some form of mitigating action such as shutting down the sound source is then necessary. However, in general it is difficult to ensure the absence of marine mammals before transmitting, so it is desirable to look for forms of sonar waveform that are potentially less harmful to marine life. One way this might be achieved is to use signals derived from natural sounds such as the vocalisations of the animals themselves. It might be expected that such sounds would appear more familiar, thus reducing possible abnormal behavioural impacts. This paper reviews the use of such waveforms and presents a preliminary estimate of their performance in practical sonar systems, along with an assessment of the potential impacts on marine life.

10:50

1aEA6. Measuring the covertness of broadband sonar waveforms. Joonho D. Park and John F. Doherty (Electrical Engineering, Pennsylvania State University, State College, PA 16802, jdp971@psu.edu)

In underwater environment, the platform uses active sonar to estimate the range to the target, covertly. The target employs detectors designed to find anomalies in its ambient environment, including man-made waveforms such as ones transmitted by the platform. However, the structure of the waveform is unknown. In this scenario, we try to measure the covertness of the waveforms transmitted by the platform in various scenarios using a quantity related to relative entropy, and the performance of range estimation using the covert waveform. At the target, we measure the maximum probability of detection of the sonar waveform, for a specified false alarm rate. At the platform, we measure the probability of estimating the range to the target correctly. The performances of both processes are dependent on the amount of information one has about the ambient environment, the structure of the transmitted waveform, and the actual range between the platform and the target. We show how the performances are dependent on the accuracy of the knowledge about these elements.

11:15

1aEA7. Considerations in designing piezocomposite transducers and arrays for broadband sonar systems. Barry Doust, Tim Mudarri, Connie Ursch, Joe Aghia, and Brian Pazol (Electroacoustics, Materials Systems Inc., 543 Great Rd., Littleton, MA 01460, bdoust@matsysinc.com)

The term broadband is commonly used to refer to the capability of a sonar system. Recent advances in signal processing and system electronics have re-defined capabilities for these systems and introduced new complex broadband pulse requirements for the sonar transducer. Advances in 1-3 piezocomposite technology have addressed this need through optimization of materials and innovative electroacoustic designs to provide high performance broadband solutions. Taking full advantage of this transducer technology requires careful consideration of the total system performance including dynamic range of receiver electronics, available transmit voltage/current/power, directivity and array configuration. This generalized study considers the design options available for optimizing transducers and arrays for broadband operation. Several Langevin or sandwich style Piezocomposite transducer configurations will be presented for both monostatic (two-way) and bistatic (separate receive and transmit) transducer systems and the trades in terms of total system performance. The study includes comparison of design optimization methodologies based on peak frequency response, impedance phase center or maximum transmit power factor, and combined two way system response. Emphasis will be on conventional piezoceramic materials with some discussion of second generation single crystal materials and their potential in future systems.

Contributed Paper

11:40

1aEA8. A generalized sinusoidal frequency modulated waveform for active sonar. David A. Hague and John R. Buck (Electrical and Computer Engineering, University of Massachusetts Dartmouth, North Dartmouth, MA 02747, david.a.hague@gmail.com)

Pulse-Compression or Frequency Modulated (FM) active sonar waveforms provide a significant improvement in range resolution and reverberation suppression over Continuous Wave (CW) waveforms. The Sinusoidal FM (SFM) waveform modulates its instantaneous frequency (IF) by a sinusoid to achieve high Doppler sensitivity while maintaining desirable reverberation suppression. This allows the SFM waveform to resolve target velocities much better than the Doppler tolerant Hyperbolic FM waveform. However, the SFM suffers from poor range resolution as the Auto-Correlation Function (ACF) contains many ambiguous peaks generated by the periodicity of the SFM's IF. The periodic sidelobes in the ACF for the SFM signal are similar to those exhibited by periodic CW waveforms, which motivated the development of FM waveforms to improve range resolution. This suggests that modifying the SFM waveform to use an aperiodic modulating function should improve range resolution while preserving Doppler sensitivity. This talk presents an active sonar waveform where the IF function is itself an FM chirp waveform, and for which the SFM is a special case. This generalized sinusoidal FM waveform resolves target range and velocity with reverberation suppression comparable to other well-established FM waveforms. [Work supported by ONR and the SMART Program.]

MONDAY MORNING, 22 OCTOBER 2012 — ANDY KIRK A/B, 10:00 A.M. TO 12:00 NOON

Session 1aMU

Musical Acoustics: General Topics in Musical Acoustics

Thomas R. Moore, Chair
Department of Physics, Rollins College, Winter Park, FL 32789

Contributed Papers

10:00

1aMU1. Choir hearing responses: Rehearsal versus performance configurations. Glenn E. Sweitzer (Sweitzer LLP, 4504 N Hereford Dr, Muncie, IN 47304, glenn.sweitzer@gmail.com)

Choir member responses to hearing (sung parts) in a rehearsal room are compared with those for its associated performance stage. Anonymous scaled responses from each choir member are gathered simultaneously using a personal response system. The protocol is repeated in a rehearsal room for the choir voices configured by 1) part versus mixed; and 2) on risers versus flat floor. On the performance stage, the protocol is repeated for same. Prior to each set of responses, the choir sings a prayer familiar to the choir members. The responses vary widely between rehearsal and performance venues, and by configuration in each. These findings suggest that choir member response may be largely ignored in the design and operation of choir rehearsal and performance facilities. Potentials for improving choir member hearing in existing rehearsal and performance venues is discussed.

10:15

1aMU2. The origins of longitudinal waves in piano strings. Brandon August, Nikki Etchenique, and Thomas R. Moore (Department of Physics, Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

The importance of longitudinal waves in piano strings has been previously identified by several investigators. Recent experimental work has provided insight into the origin of these waves and their relationship to the transverse string motion. These measurements indicate that there are multiple regimes in which longitudinal waves are created through different processes.

10:30

1aMU3. Automatic transcription of monophonic piano music. Fatemeh Pishdadian and Jill K. Nelson (Electrical and Computer Engineering, George Mason University, 4400 University Drive, ECE Department, Volgenau School of Engineering, Fairfax, VA 22030-4444, fpishdad@masonlive.gmu.edu)

Automatic music transcription refers to the process of transforming an acoustic musical signal into a written symbolic representation, e.g. a score. This process consists of extracting the parameters of note events, for instance pitches, onset times, and durations, from a raw acoustic signal. We have developed a novel algorithm for transcription of monophonic piano music, which addresses the challenges of pitch and note sequence detection in two stages: (1) The K-Nearest Neighbor (KNN) classification method is employed to identify K pitch candidates, based on spectral information, for each note event. (2) The most likely note sequence is determined by running a best-first tree search over the note candidates based on both spectral information and note transition probability distributions. The proposed two-step approach provides the performance gain achieved by incorporating note transition probabilities while maintaining significantly lower computational complexity than existing support vector machine and hidden Markov model methods. The algorithm was evaluated on a database comprised of excerpts from Bach's inventions. By performing a low complexity tree search based on note transition information, we achieve approximately 10% improvement over using only spectral information, correctly classifying roughly 85% of the notes in the database.

10:45

1aMU4. Multiple-timbre fundamental frequency tracking using an instrument spectrum library. Mert Bay and James W. Beauchamp (Electrical & Computer Engineering, University of Illinois at Urbana-Champaign, Champaign, IL 61820, mertbay@illinois.edu)

Recently many researchers have attempted automatic pitch estimation of polyphonic music (e.g., Li et al., IEEE Trans ASLP, 2009). Most of these attempts have concerned themselves with the estimation of individual pitches (F0s) while not associating the estimated pitches with the particular instruments that produce them. Estimating pitches for each instrument will lead to full music transcription. Individual instrument F0 tracks can be used in music information retrieval systems to better organize and search music. We propose a method to estimate the F0 tracks for a set of harmonic instruments in a sound mixture, using probabilistic latent component analysis (PLCA) and collections of basis spectra indexed by F0 and instrument learned in advance. The PLCA model is extended hierarchically to explain the observed input mixture spectra as a sum of basis spectra from note(s) of various instruments. The polyphonic pitch tracking problem is posed as inferring the most likely combination of the active note(s) from different instruments. Continuity and sparsity constraints are enforced to better model how the music is produced. The method was trained on a common instrument spectrum library an evaluated using an established polyphonic audio dataset.

11:00

1aMU5. Absolute pitch is associated with a large auditory digit span: A clue to its genesis. Diana Deutsch and Kevin Dooley (Department of Psychology, University of California, San Diego, 9500 Gilman Drive, La Jolla, CA 92093-0109, ddeutsch@ucsd.edu)

Absolute pitch (AP) is very rare in North America and Europe, and its genesis is unclear. Its prevalence is far higher among tone language speakers, and among those with early onset of musical training. However, most nontone language speakers with early and extensive musical training do not possess AP. To test the hypothesis that an unusually large auditory memory is involved in the genesis of AP, at least in nontone language speakers, we recruited 7 AP possessors, and 20 AP nonpossessors. All subjects were primary speakers of English, had begun musical training at $\leq$ age 6, and were UCSD students or recent graduates. The two groups were matched for age and years of musical experience. All subjects were administered an auditory digit span test, followed by a visual digit span test, with digits presented 1/sec. While the average auditory digit span was 8.1 digits for the AP nonpossessors, it was 10.0 digits for the AP possessors. This difference between the two groups was highly significant ($p = 0.0015$, 1-tailed). The AP possessors also marginally outperformed the nonpossessors on the visual digit span test; however this difference was nonsignificant. These new findings provide a clue to a genetic component of AP.

11:15

1aMU6. Perception of musical and lexical tones by musicians and nonmusicians. Chao-Yang Lee and Allison Lekich (Communication Sciences and Disorders, Ohio University, Grover W225, Athens, OH 45701, leec1@ohio.edu)

This study explores the relationship between musical and linguistic pitch perception. We asked whether the ability to identify musical tones is associated with the ability to identify lexical tones. English-speaking musicians and nonmusicians were asked to identify Taiwanese level tones produced by multiple speakers. Because pitch range varies across speakers and the tones were produced in isolation, participants had to estimate relative pitch height without cues typically available for speaker normalization. The musician participants were also asked to identify synthesized musical tones without a reference pitch. The results showed that both musicians and nonmusicians were able to identify Taiwanese tones above chance, but only for tones in the extremes of the speakers' overall pitch range. Preliminary data from the musicians show that musical tone identification accuracy was low and not associated with accuracy in the Taiwanese tone task. Implications of these findings for the music-speech relationship are discussed.

11:30

1aMU7. Measurement and analysis of timing asynchronies in ensemble performance. Gang Ren, Stephen Roessner, Samarth Hosakere Shivaswamy (Dept. of Electrical and Computer Engineering, Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu), Dave Headlam, and Mark Bocko (Dept. of Electrical and Computer Engineering; Dept. of Music Theory, Eastman Sch. of Music, Univ. of Rochester, Rochester, NY)

Timing asynchrony is an important timing descriptor of ensemble music performance. In this paper the timing asynchrony is measured as offsets between "concurring" music onsets. Specifically we measure the offset between music note onsets that are prescribed to be synchronized according to the music score. First, we conduct measurements on the multi-track audio with separate instrument tracks. These multi-track materials are recorded by using acoustically isolated recording booth or by conducting multiple recording rounds. We then perform statistical analysis both on individual asynchrony points and on asynchrony points at key coordination locations, such as the start and the end of a music phrase. We emphasize in the analysis that the musical timing asynchronies should not be treated only as performance discrepancies because part of these micro-deviation patterns provide artistic "lively" elements. We also generalize the proposed framework to mixed-down polyphonic recordings. For polyphonic recordings where clear onsets can be identified we perform similar measurement and analysis algorithms as for separated multi-track recordings. For mixed-down polyphonic recordings where a clear separation of instrument tracks is not possible, we use onset dispersions, which measure the offset range of sonic partials onsets in an coordination points, as an alternative timing asynchrony descriptor.

11:45

1aMU8. The implementation of pyschoacoustical signal parameters in the wavelet domain. Matt Borland and Stephen Birkett (SYDE, University of Waterloo, 3-98 John St. W., Waterloo, ON N2L1C1, Canada, mjborlan@uwaterloo.ca)

Introducing Wavelet techniques into the psychoacoustical analysis of sound signals provides a powerful alternative to standard Fourier methods. In this paper the reformulation of existing psychoacoustical signal parameters using wavelet methods will be explored. A major motivation for this work is that traditional psychoacoustical signal analysis relies heavily on the Fourier transform to provide a frequency content representation of a time signal, but this frequency domain representation is not always accurate; especially for sounds with impactive components. These impactive events can have a more significant contribution to the calculation of psychoacoustical signal parameters by reformulating existing psychoacoustic parameters in the Wavelet domain that are dependent on the Fourier transform. To provide concrete examples of the results simulated "plucked string" sounds are analyzed with analogous Fourier and Wavelet domain signal parameters to demonstrate the difference in performance achieved using Wavelet methods for sounds which have impactive components.

Session 1aNS

Noise, Architectural Acoustics and Physical Acoustics: Sound Absorption of Micro-Perforated Structures

Li Cheng, Cochair
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Ning Xiang, Cochair
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Chair's Introduction—8:55

Invited Papers

9:00

1aNS1. On the use of micro-perforates for machinery and vehicle noise control. Mats Åbom and Sabry Allam (The Marcus Wallenberg Laboratory, KTH-The Royal Inst of Technology, Stockholm 10044, Sweden, matsabom@kth.se)

A micro-perforated plate (MPP) is a perforated plate with holes typically in the sub-millimeter range and perforation ratio around 1%. The values are typical for applications in air at standard temperature and pressure (STP). The underlying acoustic principle is simple: To create a surface with a built in damping that effectively dissipates sound waves. To achieve this, the specific acoustic impedance of a MPP is normally tuned to be of the order of the characteristic wave impedance in the medium (400 Pa*s/m in air at STP). The traditional application for MPP's has been building acoustics, normally in the form of a so called panel absorber to create an absorption peak at a selected frequency. However, MPP's made of metal are also well suited for machinery and vehicle noise control. For instance MPP's have the potential to be used instead of porous materials in dissipative mufflers, which not only can save weight but also offer a non-fibrous alternative. Furthermore, since MPP's have a large steady flow resistance they can be used as acoustically absorbing guide vanes at duct bends or as a fan housing. One important issue for these applications is the effect of flow on the MPP impedance. This issue plus a number of applications related to vehicle noise, have been studied at KTH during the last decade and this paper aims at summarizing the main results.

9:20

1aNS2. The effect of flexibility on the acoustical performance of microperforated materials. J. S. Bolton (Ray W. Herrick Laboratories, School of Mechanical Engineering, Purdue University, 140 S. Martin Jischke Drive, West Lafayette, IN 47907-2031, bolton@purdue.edu)

In conventional models of microperforated materials, the solid layer in which the holes are formed is usually considered to be rigid. However, microperforated materials are often thin, say less than 0.5 mm in thickness, and are sometimes made of lightweight polymeric materials. Experimental measurements suggest that when the mass per unit area of a microperforated material is less than approximately 0.5 kg per square meter, the motion of the solid layer becomes important. The solid layer is driven into motion both by the incident pressure acting on its surface and by viscous forces generated within the perforations. The ability of a microperforate to dissipate acoustical energy depends on there being relative motion between the air in the perforations and the solid layer: motion of the solid may either help or hurt this effect, particularly when the solid layer is supported on a grid-like structure, since the individual segments of the microperforate then exhibit modal behavior. In this presentation, models of this behavior will be described, and examples will be given in which essentially membrane-like behavior is modified by the presence of the perforations, and conversely, in which essentially rigid microperforated layer behavior is modified by vibration of the solid layer.

9:40

1aNS3. Micro-perforated elements in complex vibro-acoustic environment: Modelling and applications. Xiang Yu (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hong Kong SAR, China), Laurent Maxit, Jean-Louis Guyader (Laboratoire Vibrations Acoustique, Institut National des Sciences Appliquées (INSA) de Lyon, Lyon, France), and Li Cheng (Department of Mechanical Engineering, The Hong Kong Polytechnic University, FG611, Hung Hom, Kowloon, Hong Kong SAR 999077, China, mmlcheng@inet.polyu.edu.hk)

Micro-perforated structures/panels (MPP) are widely used in various architectural, industrial and environmental applications for providing efficient sound absorptions. More recently, they found their use in compact mechanical systems, in which the property of the MPP is shown to be strongly influenced by the surrounding vibro-acoustic environment, which is drastically different from the one dimensional Kundt tube configuration, usually used in existing works. Unfortunately, very little has been done in this regard, due to the fact that modeling such a vibro-coustic system with MPPs as integrative elements is a very challenging task, not even to mention the optimization. Recently, a vibro-acoustic formulation based on the Patch Transfer Function (PTF) approach was proposed to model micro-perforated structures in a complex vibro-acoustic environment. As a sub-structuring approach, PTF allows assembling different vibro-acoustic subsystems, including micro-perforations and the flexibility of a MPP, through coupled surfaces. The proposed formulation provides explicit representation of the coupling among subsystems, with enhanced capability of handling system complexities and facilitating system optimization. In this talk, the overall approach will be reviewed and applied to a number of typical examples. The versatility and efficiency of the method as well as the underlying physics are demonstrated.

10:00–10:30 Break

10:30

1aNS4. A straightforward method toward efficient and precise impedance measurement for microperforation panels under flow conditions. Xiaodong Jing (School of Jet Propulsion, Beijing University of Aeronautics and Astronautics, No. 37, Xueyuan Road, Beijing 100191, China, jingxd@buaa.edu.cn)

This paper addresses the problem how to measure acoustic impedance of microperforation panels (MPPs) under flow conditions. Since 1960s, many different methods have been proposed to tackle this problem, mainly motivated by the aim of reducing noise emission from aeroengines, ventilators and other fluid machines. It has been found that the presence of flow favorably enhances the damping of MPPs due to the mechanism of sound-vortex interaction. This, however, leads to flow-dependent acoustic impedance whose determination is rather difficult. Despite considerable efforts over the past decades, there is still stringent need for developing efficient and precise impedance measurement method under flow conditions in order to fully explore the potentials of MPPs. Towards this goal, a straightforward method has been put forward to measure the acoustic impedance of an MPP lined in a flow duct (JASA, 124(1), 227-234). The basic principle is that the dominate axial wavenumber is extracted from the measured wall sound pressure by means of Prony method, thereby the unknown acoustic impedance is algebraically solved from the dispersion equation. In this paper, the straightforward method is further extended to incorporate the effects of flow boundary layer and higher-order acoustic modes that are essentially important for practical applications.

10:50

1aNS5. Hybrid silencer by using micro-perforated plate with side-branch cavities. Xiaonan Wang, Yatsze Choy, and Li Cheng (Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hong Kong 852, mmyschoy@polyu.edu.hk)

A plate silencer consists of an expansion chamber with two side-branch cavities covered by light but extremely stiff plates. It works effectively with wide stopband from low-to-medium frequencies only if the plate is extremely stiff, to ensure a strong reflection of acoustic wave to the upstream in the duct. However, a plate with slightly weak bending stiffness will result in non-uniform transmission loss (TL) spectra with narrowed stopband. In this study, a hybrid silencer is proposed by introducing micro-perforations into the plate to elicit the sound absorption in order to compensate for the deficiency in the passband caused by the insufficient sound reflection in certain frequency range due to plate with weaker stiffness. A theoretical model, capable of dealing with the strong coupling between the vibrating micro-perforated plate and sound fields inside the cavity and the duct, is developed. Through a proper balancing between the sound absorption and reflection, the proposed hybrid plate silencer with moderately stiff plates is shown to outperform the typical plate silencer with very stiff plate. Whilst releasing the harsh requirement on the bending stiffness of the plate, the proposed hybrid silencer provides a more flattened and uniform TL and a widened stopband by about 30%.

Session 1aPA

Physical Acoustics, Architectural Acoustics, and Noise: Recent Developments in Computational Acoustics for Complex Indoor and Outdoor Spaces

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Computer Science, University of North Carolina at Chapel Hill, Chapel Hill, NC 27599-3175

Chair's Introduction—8:00

Invited Papers

8:05

1aPA1. Modeling reflections from rough surfaces in complex spaces. Samuel Siltanen, Alex Southern, and Lauri Savioja (Department of Media Technology, Aalto University, PO Box 15400, Aalto FI-00076, Finland, Lauri.Savioja@aalto.fi)

Acoustic reflections from rough surfaces occur both in indoor and outdoor environments. Detailed modeling of such reflections is possible with wave-based modeling algorithms. However, their time and resource consumption limits the applicability of such algorithms in practical modeling tasks where the modeled space is large or even unbounded. On the other hand, geometrical acoustics modeling techniques are more efficient in most cases, but they are not able to capture the wave interaction with the rough surface. The presented solution combines a geometric modeling algorithm with a theoretical model of the rough surface. The geometric algorithm is an efficient beam tracer. The theoretical model assumes long wavelength compared to the dimensions of the surface details. The model shows that the effect of the rough surface can be approximated with a exponential decay tail after an impulse in the time domain impulse response. A further approximation is to present multiple such reflections with a decay resembling a gamma distribution. Several complex example cases with a large number of reflections are shown. In addition, comparison of the theoretical model to finite-difference time-domain algorithm modeling results is given. The results support the applicability of the presented approach to practical modeling tasks.

8:25

1aPA2. Computational modeling of broadband acoustic fields in enclosures with specular reflection boundaries using a first-principles energy method. Donald B. Bliss, Krista A. Michalis, and Linda P. Franzoni (Mechanical Engineering, Duke University, Durham, NC 27708, dbb@duke.edu)

Steady-state sound fields in enclosures with specular reflection boundaries are modeled with a first-principle energy-intensity boundary element method using uncorrelated broadband directional sources. The specular reflection field is represented by a limited set of spherical harmonics, orthogonal on the half-space. For each boundary element, the amplitudes of these harmonics are determined from the incident field from all other elements and sources, and are subject to an energy conservation integral constraint using a Lagrange multiplier method. The computational problem is solved using an iterative relaxation method starting from the 3-D diffuse reflection solution. At each iteration, directivity harmonics are estimated by post-processing and the influence matrix is refined accordingly. For internal sources, simple first reflection images improve accuracy with virtually no penalty on computation time. Convergence occurs in relatively few relaxation steps. Extrapolating to an infinite number of boundary elements and iterations gives very accurate results. Results are compared to exact benchmark solutions obtained from a frequency-by-frequency modal analysis, and to a broadband image method. The method of absorption scaling is verified for 3-D cases, and showing that the spatial variation in rooms is largely determined by source position and the relative distribution of absorption, but not the overall absorption level.

8:45

1aPA3. An edge-source integral equation for the calculation of scattering. U. Peter Svensson (Acoustics Research Centre, Department of Electronics and Telecommunications, Norwegian University of Science and Technology, NTNU IET, Trondheim - NO-7491, Norway, svensson@iet.ntnu.no) and Andreas Asheim (Department of Computer Science, Katholieke Universiteit, Heverlee, Belgium)

A new integral equation for the scattering from rigid, or pressure-release, convex polyhedra and disks is presented. It is based on edge diffraction as a complement to the geometrical acoustics components, and uses directional edge sources as unknowns in the frequency-domain integral equation. Comparisons with reference results show that the new method gives correct results down to 0 Hz in spite of being a geometrically based method [Asheim & Svensson, Report TW610, KU Leuven, Dept. of Computer Science, 2012]. The two-dimensional unknowns are solved for all edges, straight or curved, of the scattering object, but this reduces to a one-dimensional unknown for certain geometries such as axisymmetric scattering from a circular thin disk, or plane-wave incidence onto a polygonal cylinder. The general integral equation can be solved with the regular Nyström technique, iteratively or by direct inversion. The scattered field is computed in a post-processing stage, and is added to the geometrical acoustics and first-order diffraction components, which are computed separately. The formulations for non-convex external, as well as internal, geometries are laid out, and time-domain versions of the integral equation are linked to previously published work on edge diffraction impulse responses.

9:05

1aPA4. A fast wave-based hybrid method for interactive acoustic simulation in large and complex environments. Tian Xiao (EE Boost Inc., 618 Powers Ferry RD, Cary, NC 27519, xiao@eeboost.com)

Modern acoustic applications require fast and accurate simulation in large and complex environments. Current methods are rather limited to very simple environments or not able to perform fast interactive simulations. Our objective is to develop an innovative and practical method to perform real-time or near real-time accurate interactive simulation in large and complex environments, such as urban environments up to one kilometer. A new method, the embedded hybrid method using immersed boundary within k-space PSTD has been proposed and developed to achieve the objective. Its performance, accuracy, parallel hardware acceleration, and capabilities to handle all kinds of environmental complexities have been rigorously validated and verified by a number of examples. It shows that with the utilization of modern many-core GPUs, the method can perform real-time or near real-time accurate interactive simulation in large environments of hundreds to thousands of meters with all kinds of complexities including (1) inhomogeneous and absorptive medium, (2) curved and arbitrarily-shaped objects, (3) and moving sources, receivers, and objects, and time-varying medium.

9:25

1aPA5. Adaptive rectangular decomposition: A spectral, domain-decomposition approach for fast wave solution on complex scenes. Nikunj Raghuvanshi (Microsoft Research, 1 Microsoft Way, Redmond, WA 98052, nikunjr@gmail.com), Ravish Mehra, Dinesh Manocha, and Ming C. Lin (Computer Science, University of North Carolina at Chapel Hill, Chapel Hill, Washington)

Computational wave propagation is increasingly becoming a practical tool for acoustic prediction in indoor and outdoor spaces, with applications ranging from noise control to architectural acoustics. We discuss Adaptive Rectangular Decomposition (ARD), that decomposes a complex 3D domain into a set of disjoint rectangular partitions. Assuming spatially-invariant speed of sound, spectral basis functions are derived from the analytic solution and used to time-step the field with high spatio-temporal accuracy within each partition. This allows close-to-Nyquist numerical grids, with as low as 3 points per wavelength, resulting in large performance gains of ten to hundred times compared to the Finite-Difference Time-Domain method. The coarser simulation grid also allows much larger computational domains. ARD employs finite-difference interface operators to transfer waves between adjoining rectangular partitions. We show that efficient, spatially-compact interface operators can be designed to ensure low numerical errors. Numerical solutions obtained with ARD are compared to analytical solutions on simple geometries and good agreement is observed.

9:45

1aPA6. Real-time sound propagation and noise modeling in outdoor environments using Equivalent Source Formulation. Ravish Mehra, Dinesh Manocha, Lakulish Antani (Computer Science, University of North Carolina at Chapel Hill, Columbia Street, Chapel Hill, NC 27599-3175, dmanocha@gmail.com), and Nikunj Raghuvanshi (Microsoft Research, Microsoft, Redmond, WA)

We address the problem of wave-based sound propagation in outdoor and urban environments. The goal is to accurately simulate acoustic effects, including interference, diffraction, scattering, and higher-order wave effects in large outdoor scenes. We give an overview of a precomputed wave-based solver that is based on equivalent source method and is mainly applicable to large, open scenes [Mehra et al. 2012]. As part of a preprocessing step, it computes a per-object transfer function that models the scattering behavior of each object, and handles pair-wise acoustic coupling between objects using inter-object transfer functions. The runtime component involves fast summation over all outgoing equivalent sources for all objects at the listener location. We highlight its runtime performance and memory efficiency and use it for noise modeling and prediction in outdoor scenes spanning a few hundreds of meters. The sound field is computed in three dimensions, modeling frequency-dependent propagation above ceilings, around buildings and corners, and high-order interactions.

10:05–10:25 Break

Contributed Papers

10:25

1aPA7. Coupling of parabolic equation method with the scattering of sound. Santosh Parakkal, D. Keith Wilson, and Sergey N. Vecherin (US Army, ERDC-CRREL-NH, 72 Lyme Road, Hanover, NH 03755, Santosh.Parakkal@usace.army.mil)

The problem of sound scattering by an infinitely long penetrable and impenetrable cylinder suspended over a realistic impedance ground is investigated. The analytical approach using the image source method in the scattering of sound involves expressing the total sound field at any receiver point (over a locally reacting ground) as the sum total of the direct field, the ground reflection from the source, scattered field by the actual cylinder and finally by its image. The exact solutions can then be expressed as an infinite series, containing Bessel and Hankel functions of increasing order. The coefficients of the scattered field are determined by matching the desired boundary condition (for a rigid circular cylinder, for example, the normal component of velocity is zero on the boundary). Although the preceding approach is commonly used in theoretical treatments of sound scattering, to the best of our knowledge this is the first time it has been attempted numerically with coupling to the Parabolic Equation (PE) method. Presented is a Two-dimensional case of sound scattering of PE generated acoustic field by an impenetrable (soft or rigid) and penetrable cylindrical obstacle over a finite impedance ground in a non-refractive atmosphere.

10:40

1aPA8. Time-domain simulation of long-range sound propagation in an atmosphere with temperature gradient. Z. C. Zheng and Guoyi Ke (Aerospace Engineering, University of Kansas, 1530 W 15th Street, Lawrence, KS 66045, zzheng@ku.edu)

A numerical model for linearized Euler equations using finite difference in time-domain (FDTD) simulation is developed to simulate sound propagation with temperature gradient in the atmosphere. The speed of sound in the air varies with the temperature at different altitude above the ground due to the effect of temperature gradient. For sound propagation at long ranges, an algorithm of moving-frame method is implemented with parallel computation. The numerical results are compared with analytical solutions for sound propagation with downward and upward refraction caused by the speed of sound linearly increasing (downward refraction) or decreasing (upward refraction) with altitude. The 2D normal mode analytical solutions are used

to compare the downward refraction results, and the residue series analytical solutions are used to compare the upward refraction results. The comparison show that the numerical simulation results agree very well with the analytical solutions for both downward refraction and upward refraction cases. Several examples of long- and short-range simulation results are then presented.

10:55

1aPA9. Wine glass resonance experiment and demonstration. Benjamin C. Thines (University of Central Arkansas, Conway, AR 72034, thinesbc@gmail.com)

Breaking a wine glass with sound is a visually striking achievement and a great way to get potential students interested in Physics. The goal of this project is to not only break the wine glass but to build an apparatus that is portable and easily setup for lecture room demonstrations as well as outreach to area schools. The apparatus should also provide enough visibility for a room full of observers to easily see the process. In order to be able to observe the small deflections of the object a variable frequency strobe will be employed. A strobe has the benefit of being able to see in real time what is going on at a much higher frequency than the human eye would normally perceive. In a larger setting a camera could be used to relay the relatively small image of the wine glass to a projector for better visibility. From a more technical stand point, the project will provide an opportunity to experiment with resonance on a variety of different shapes and compositions of items. In order to prepare for the final demonstration, many different wine glasses will be tested in the test chamber.

11:10

1aPA10. Aperiodicity and ground effects on the sonic crystal noise barriers. Shahram Taherzadeh, Imran Bashir, Keith Attenborough, and Alvin Y. Chong (MCT, The Open University, Walton Hall, Milton Keynes, Bucks MK7 6AA, United Kingdom, s.taherzadeh@open.ac.uk)

Sonic crystal structures consisting of periodically-arranged solid vertical cylinders can act as sound barriers at certain frequencies. Their performance depends on the filling fraction which is determined by the spacing and cylinder radius. To be effective the filling fraction must be high. This means that periodic arrays with relatively low filling fractions such as in trees belts are not effective as traffic noise barriers. The effects of partially perturbing the positions of sonic crystal elements have been investigated by modelling and laboratory measurements and have been shown to improve the insertion loss of the periodic structure. It is argued that partial perturbation of regular tree planting near highways will improve their noise attenuation. Furthermore, much previous research assumes the sonic crystal structure to be in the free field, i.e. no account has been taken of the presence of the ground surface. With a conventional, wall type, barrier the ground effect is reduced by presence of the barrier. Laboratory measurements have been made of periodic and aperiodic arrays of cylinders placed with their axes normal to acoustically hard and soft surfaces. It is found that the ground effects and sonic crystal band gap effects are additive.

11:25

1aPA11. Partial field decomposition of jet noise using optimally located virtual reference microphones. Alan T. Wall, Kent L. Gee, Tracianne B. Neilsen (Dept. of Physics and Astronomy, Brigham Young University, N283 ESC, Provo, UT 84602, alantwall@gmail.com), and Michael M. James (Blue Ridge Research and Consulting, Asheville, NC)

The application of partial field decomposition (PFD) techniques based on a singular value decomposition to jet noise fields is useful for estimating the number of incoherent (equivalent) noise sources within a jet and for implementing near-field acoustical holography, but it does not generally provide physically meaningful partial fields (i.e. partial fields related to individual sources). Among several PFD methods that were designed to generate physically meaningful partial fields, the method developed by Kim et al. [JASA 115(4), 2004] finds the optimal locations of references in a sound field and places virtual references at those locations. In past investigations this method has been successfully applied to locate discrete numerical and physical sources and to generate partial fields related to each source. In this study, Kim's method is applied to a full-scale jet installed on a military aircraft in an attempt to obtain physically meaningful partial fields. The partial fields obtained using these optimally located virtual references are compared to the partial fields obtained from other PFD methods.

11:40

1aPA12. Observations with grazing and vertical incidence methods of ground impedance estimation. Michael J. White (US Army ERDC/CERL, PO Box 9005, Champaign, IL 61826, michael.j.white@usace.army.mil), George W. Swenson, and Jeffrey D. Borth (Department of Electrical and Computing Engineering, University of Illinois at Urbana-Champaign, Urbana, IL)

At locations near a ground surface of interest, the ground impedance may be evaluated using a loudspeaker suitably disposed to broadcast toward both the ground and two vertically-separated microphones. By evaluating the complex gain ratio between the pair for a single tone, the usual approximation to the Green function for the monopole field above a locally-reacting ground can be inverted to find the surface impedance. This inversion is somewhat sensitive to noise, but it can also be found to vary according to the placement of microphones and speaker, apart from speaker directivity. Recently the method of Soh et al. [Soh et al. JASA 128:5 EL286 2010] was proposed for measuring ground impedance some distance from the source. Because the method relies more directly on the boundary condition at the ground, it may offer some benefit for use at shorter distances as well. We discuss the comparisons between measurements interpreted by both techniques, consider noise entering the estimation process and placement effects.

Session 1aSA

Structural Acoustics and Vibration: Damping Applications and Modeling

Benjamin M. Shafer, Chair
Building Acoustics, Conestoga-Rovers & Associates, Inc., 1117 Tacoma Ave., Tacoma, WA 98402

Invited Papers

9:00

1aSA1. Damping: Some often overlooked facts. Eric E. Ungar (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138-1118, eungar@acentech.com)

Damping - dissipation of mechanical energy - has significant effects on only some types of vibrations. Damping can result from many mechanisms, many of which cannot readily be modeled, but prediction of details of motions requires correct representations of the dominant mechanisms. The assumption of viscous damping permits one to analyze vibrations via relatively easily solved linear differential equations, but can lead to results that do not represent reality. Several measures of damping are based on simple models involving frequency-independent viscous damping, but realistic damping behavior often may be better represented by a frequency-dependent loss factor. The chemical properties of plastics and elastomers generally are not known well enough to permit assessment of the behavior of such materials without dynamic measurements. Structural configurations with relatively high damping may be obtained by combining high-damping polymeric materials with efficient structural materials only if the configurations are such that for a given deformation the high-damping material stores a considerable fraction of the total mechanical energy. This is manifest in the behavior of free-layer and constrained-layer damping treatments and in their design equations, which also indicate that a damping material that is very good for one of these types of treatments may not be good for the other.

9:25

1aSA2. Acoustical performance of damped gypsum board in double wood stud wall assemblies. John LoVerde and David W. Dong (Veneklasen Associates, 1711 16th St, Santa Monica, CA 90404, jloverde@veneklasen.com)

A common assembly for demising walls in multifamily residential projects is double wood studs with multiple layers of gypsum board on one or both sides. While improving the acoustical performance, installing multiple layers of gypsum board adds significant cost to the project; complicates scheduling, materials storage, and delivery; and may require additional inspections. On some types of projects, removing these complications has been desired. In these cases, the use of damped gypsum board has been pursued to reduce the number of layers of gypsum board on the demising wall while attempting to achieve equivalent acoustical performance. Acoustical testing was performed to determine if damped gypsum board was a suitable alternative to the standard gypsum board assemblies in typical demising constructions. Laboratory and field tests were performed on double wood stud walls with a variety of gypsum board configurations, and the results are presented.

9:50

1aSA3. Effects of boundary conditions on the transmission of impulsive sound at low frequencies through building components into enclosed spaces. Marcel C. Remillieux (Mechanical Engineering, Virginia Tech, 149 Durham Hall, Blacksburg, VA 24061, mremilli@vt.edu)

The transmission of impulsive sound at low frequencies, through building components, into enclosed spaces is solved numerically. This investigation is directed at what is essentially a transient problem. The shapes of the vibro-acoustic waveforms, in particular peak values, are of principal concern since they relate directly to the auditory response of occupants and possibility to structural damage. The case of an N-wave impinging upon a window panel backed by a rigid rectangular enclosure is considered in this study. At low frequencies, the vibro-acoustic response of a typical building component is dominated by a few modes. Besides, the response is very sensitive to boundary conditions. Therefore, the vibro-acoustic response of the system can be altered by tuning the boundary conditions of the structural components. Three types of boundary conditions are examined: simply-supported, fixed, and visco-elastic. It is demonstrated that the peak amplitudes of the vibro-acoustic waveforms can be reduced significantly by the appropriate choice of boundary conditions.

10:15–10:30 Break

10:30

1aSA4. Predicting structural and acoustic-radiation loss factors using experimental modal analysis. Micah R. Shepherd, Stephen A. Hambric, and John B. Fahnline (Applied Research Lab, Penn State University, PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

Standard damping measurements cannot typically distinguish between structural losses and losses due to acoustic radiation. Since the trend in aerospace engineering is to use lighter and stiffer materials, aerospace panels often have their critical frequency at low to mid frequencies making the acoustic losses of comparable or greater value than the structural loss factors. A procedure for measuring

acoustic radiation loss factors is presented based on experimental modal analysis. Experimental modal analysis was performed on a composite panel with carbon-fiber facesheets and an aluminum honeycomb core. The loss factors using the standard decay and modal methods are compared to an energy-based method based on the power injection method. The acoustic radiation loss factors were then estimated using boundary element computations of the radiated noise with the modally reconstructed velocity as input. The acoustic radiation damping was then removed from the total damping to predict the losses due to structural damping only.

Contributed Papers

10:55

1aSA5. Vibration damping mechanisms for the reduction of sting in baseball bats. Daniel A. Russell (Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802, drussell@engr.psu.edu)

When the impact between a baseball and a bat occurs outside the "sweet-spot" region, the resulting vibration in the handle often produces a stinging sensation in the hands. Pain from a poorly hit ball is primarily felt in the fleshy web between thumb and forefinger in the top (distal) hand, and also to a lesser degree in the heel of the bottom (proximal) hand, and the sensation of sting is more prevalent in aluminum bats than in wood. Several bat manufacturers have attempted to minimize handle vibration in aluminum bats through various methods including handle grips, foam injected into the hollow handle, two-piece construction joining composite handles to aluminum barrels, and the insertion of vibration absorbers in the taper region and in the knob of the handle. This paper will assess and compare the performance of several such vibration damping applications implemented in baseball bats. Experimentally measured damping rates corresponding to the bending mode shapes responsible for sting will be compared for a variety of bat designs. The effectiveness of available commercially implemented damping mechanisms will be compared using frequency response functions and time signals.

11:10

1aSA6. An approach to increase apparent damping with reduced subordinate oscillator array mass. Aldo A. Glean, Joseph F. Vignola, John A. Judge, Teresa J. Ryan (Mechanical Engineering, Center for Acoustics, Vibrations and Structures, Catholic University of America, 620 Michigan Avenue, N.E, Washington, DC 20064, 10glean@cardinalmail.cua.edu), and Patrick F. O'Malley (Mechanical Engineering, Benedictine College, Atchison, KS)

The case of a lightly damped oscillator (primary mass) adorned with a set of substantially less massive oscillators is known as a subordinate oscillator array (SOA). An SOA can function as an energy sink on the primary, extracting vibration energy from it, thus adding apparent damping to the system. Low apparent Q is achieved by increasing non-dimensional bandwidth, which is the ratio of the bandwidth to the fundamental frequency of the primary. The mass of the subordinate set required to achieve the most rapid energy transfer from the primary is proportional to non-dimensional bandwidth squared. We have shown the limit of apparent damping achievable in these systems is the inverse of non-dimensional bandwidth. In practice, the utility of this result is limited because a great deal of mass (~30% of primary) is required to increase the apparent damping in the system such that $Q\_apparent \rightarrow 1$. This work will focus on an alternative design strategy that produces comparable increase in apparent damping with less added mass. We describe numerical optimizations in which the non-dimensional bandwidth of the isolated natural frequencies of the SOA elements and the distribution of those isolated natural frequencies are used to minimize the total mass of the SOA.

11:25

1aSA7. Viscous boundary layer correction on a pressure-field acoustic model. Yizeng Li (Department of Mechanical Engineering, University of Michigan - Ann Arbor, 2350 Hayward Avenue, Ann Arbor, MI 48109, liyizeng52@hotmail.com), Lei Cheng (BOSE Corporation, Framingham, MA), and Karl Grosh (Department of Mechanical Engineering, University of Michigan - Ann Arbor, Ann Arbor, MI)

Fluid viscosity plays an important role in many acoustics and structural acoustics problems. For example, using an inviscid approximation to the flow of fluid-loaded micro-electro-mechanical systems and micro-scale biological structures results in large errors in the predicted response. Using a linearized Navier–Stokes solution, however, increases the number of unknowns by at least a factor of three compared to an inviscid approximation where pressure is the only degree of freedom. In this work, an approximate boundary condition is developed to include fluid viscosity for coupled fluid-structure systems. The viscous effect is included as a correction term to the inviscid boundary condition, written in terms of second order in-plane derivatives of pressure. This is the key step enabling the development of a variational formulation that is directly amenable for approximation in a finite element method (FEM) code as only a minor modification to existing structural acoustic code. Hence, this approach retains the great computational advantage over the conventional viscous FEM formulation. We show results demonstrating the accuracy of the approximate boundary condition as compared to the full three dimensional Navier-Stokes solution.

Session 1pAA

Architectural Acoustics: Performing Arts Center Acoustics–Pre-Tour Talk

Robert C. Coffeen, Cochair
School of Architecture, Design & Planning, Universtiy of Kansas, Lawrence, KS 66045

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

Chair's Introduction—1:35

Invited Papers

1:40

1pAA1. Acoustical design of Calderwood Hall at the Isabella Stewart Gardner Museum. Daniel Beckmann, Motoo Komoda, and Yasuhisa Toyota (Nagata Acoustics, 2130 Sawtelle Bl., Ste. 308, Los Angeles, CA 90025, beckmann@nagata.co.jp)

The Isabella Stewart Gardner Museum opened the expansion to its historic "Fenway Court" Palace in the Back Bay district of Boston, Massachusetts in January, 2012. Renzo Piano Building Workshop was the design architect for the 70,000 square foot addition to the museum. The new Calderwood Hall serves the music programming mission of the museum by creating a 300-seat space where chamber music can equally be enjoyed by all members of the audience. A novel audience configuration is perhaps the most defining element of this highly intimate space for music, where a stage measuring 30 feet square is centered on the bottom level of the space measuring 42 feet square and 45 feet tall. Three stacked balconies, each one row deep, look down on the stage from all four sides. The integrated acoustical and architectural design features are reported.

2:00

1pAA2. Acoustical design of Muriel Kauffman Theatre at the Kauffman Center for the Performing Arts. Motoo Komoda, Kayo Kallas, Daniel Beckmann, and Yasuhisa Toyota (Nagata Acoustics, 2130 Sawtelle Bl., Ste. 308, Los Angeles, CA 90025, komoda@nagata.co.jp)

Muriel Kauffman Theatre, the horseshoe-style proscenium theater in the Kauffman Center for the Performing Arts in Kansas City, Missouri, opened in September, 2011. The 1800-seat theater was designed by the architect Moshe Safdie, and serves as the principal residence of both the Kansas City Ballet, and the Lyric Opera of Kansas City. Since ballet and opera performance were programmed as the primary use, the primary focus of the room acoustical design was to provide good natural acoustics for singers on the stage, and for the orchestra in the pit. This was achieved through careful study on the shape of the room. The walls surrounding the stage and proscenium, and the configuration of the ceiling were among the primary aspects of the room shape which were studied, as well as flexible room acoustics for programs with using amplified sound, such as pop music and spoken-word programs. The acoustical design and characteristics of the Muriel Kauffman Theatre are reported.

2:20

1pAA3. Acoustical design of Helzberg Hall at the Kauffman Center for the Performing Arts. Daniel Beckmann, Kayo Kallas, Motoo Komoda, and Yasuhisa Toyota (Nagata Acoustics, 2130 Sawtelle Bl., Ste. 308, Los Angeles, CA 90025, beckmann@nagata.co.jp)

Helzberg Hall, the arena-style concert hall in the Kauffman Center for the Performing Arts in Kansas City, Missouri opened in September, 2011. The 1600-seat concert hall was designed by the architect Moshe Safdie, and serves as the principal residence of the Kansas City Symphony. The hall measures 150 feet long, 100 feet wide, and 90 feet tall at the highest point, with an ensemble reflector suspended at 50 feet above the stage to support the stage acoustics. Stage acoustics are also enhanced by motorized risers on the stage, to give the orchestra layout a three-dimensional quality. Motorized draperies concealed in the walls also give flexibility to the acoustics of the space, in order to accommodate amplified events. A 1:10 scale model test of the hall was performed at the end of the design phase to verify the acoustics of the hall. The acoustical design and characteristics of the new space are reported.

Session 1pAB

Animal Bioacoustics, Engineering Acoustics, and Signal Processing in Acoustics: Echolocation, Bio-Sonar, and Propagation

Whitlow W. L. Au, Chair
Hawaii Institute of Marine Biology, University of Hawaii, Kaneohe, HI 96744

Contributed Papers

1:30

1pAB1. Bio-sonar signal processing. Harry A. DeFerrari and Jennifer Wylie (Applied Marine Physics, University of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, hdeferrari@rsmas.miami.edu)

An ideal sonar for biological observations in the ocean should employ very long signals for Doppler resolution, wide bandwidth for time resolution and pulse compression properties for gain in order to reduce sound pressure levels to mitigate marine mammal concerns. Signal spread in time and Doppler (leakage) must also be eliminated so that direct arrival and reverberation signals do not leak and swamp bio-signal returns when operation in a continuous transmit and receive mode. Orthogonal codes would help multi-static applications. Here two types of signals and processing are presented and analyzed; 1) M-sequences with matched filter processing and 2) Inverted binary sequence with matched- inverse processing. Ambiguity diagrams of conventional active sonar signals are compared with the two candidate signals. Then the two methods are compared by numerical simulation for propagation in shallow channels with zero Doppler returns and moving bio-scatterers. It is shown that for realistic conditions leakage from direct and zero Doppler reverberation can be eliminated not just reduced. Limits on temporal coherence times of shallow water propagations channels are discussed. These approaches improve Figure of Merit of conventional active sonar by 10 to 15 dB and displays in the time Doppler plane simplify the interpretation of clutter.

1:45

1pAB2. Effects of local shape features on the beampatterns of a dynamic biomimetic baffle. Mittu Pannala, Sajjad Z. Meymand, and Rolf Mueller (Mechanical Engineering, Virginia Tech, 1110 Washington Street, SW, Blacksburg, VA 24061, mpannala@vt.edu)

Horseshoe bats have mobile pinnae that can change their shapes as a result of active actuation. A common pattern is an alteration between an upright and a bent-tip geometrical configuration. Numerical predictions of reception beampatterns associated with these different shape geometries have indicated that the upright configurations are associated with beampatterns dominated by a single mainlobe whereas the bent-tip configurations have prominent sidelobes. Using a biomimetic baffle prototype, we have found that this effect can be reproduced qualitatively with just a plain obliquely truncated cone that is fabricated from flexible material (rubber) and bent at the tip. However, local shape features can have a strong impact on the quantitative expression of this effect. The three features studied here were a vertical ridge, an equivalent to the bat's antitragus, and a lateral incision into the baffle rim. The effects of these features on the beampatterns where found to interact with each other and also depend strongly on the deformation stage of the baffle shape. Hence the corresponding biological features may offer bats an opportunity to fine-tune their beampatterns as a function of deformation stage. However, control strategies for biomimetic devices with variable beampatterns have yet to be developed.

2:00

1pAB3. Analysis of biosonar beamwidth with spherical harmonics power spectra. Mohammad Motamedi (Mechanical Engineering, Virginia Tech, Blacksburg, VA 24060, m.motamedi.85@gmail.com), Qiuping Xu, Washington Mio (Mathematics, Florida State University, Tallahassee, FL), and Rolf Mueller (Mechanical Engineering, Virginia Tech, Blacksburg, VA)

Ultrasound emission and reception in bats are characterized by beampatterns that can be predicted from digital models of the geometries of noseleaf and pinna structures numerically. One obvious way in which beampatterns can differ across species is in the overall width (angular extent) of the beams. But since biosonar beams also vary in many other ways such as in their orientation and their shape, quantifying overall beamwidth poses a challenge. Here, the power spectrum of a beam decomposition based on spherical harmonics has been used to obtain a quantitative measure of beamwidth. Due to the reciprocity between transform domains, narrower beams correspond to spectra with a higher bandwidth and vice versa. Using power spectral magnitudes, it has been possible to classify emission and reception beampatterns as well as beampatterns from different taxonomic bat groups across 176 noseleaf and 185 pinna samples representing 106 species. Furthermore, when the power spectra of the actual biosonar beams were replaced with those of fitted single heat functions, classification performance suffered little. Since the heat kernel approximations did not differ in any property other than beamwidth, this demonstrates that beamwidth alone is a discriminating factor between bat taxonomic groups as well as emission and reception.

2:15

1pAB4. Examining the effects of propagation on perceptual features used for automatic aural classification. Carolyn M. Binder, Paul C. Hines, Sean P. Pecknold, and Jeff Scrutton (Defence R&D Canada, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, carolyn.binder@drdc-rddc.gc.ca)

A prototype aural classifier has been developed at Defence R&D Canada that uses perceptual signal features which model the features employed by the human auditory system. Previous effort has shown the classifier reduced false alarm rates and successfully discriminated cetacean vocalizations from several species. The current paper investigates the robustness of the aural features against propagation effects for two of those species - the bowhead and humpback whales. This is achieved by comparing the classification results of the original vocalizations to classification results obtained after the vocalizations were re-transmitted underwater over ranges of 2 to 10 km. To gain additional insight into the propagation effects, synthetic bowhead and humpback vocalizations, with features similar to the most important aural features for classification of bowhead and humpback vocalizations, were also transmitted. In this paper, the classifier performance on both real and synthesized vocalizations are compared before and after propagation, to quantify the effect of propagation on the features used in the aural classifier.

2:30

1pAB5. Propagation modeling techniques for marine mammal management studies. Elizabeth T. Küsel, Martin Siderius, and Scott Schecklman (Northwest Electromagnetics and Acoustics Research Laboratory, Portland State University, 1900 SW 4th Ave., Portland, OR 97201, kusele@alum.rpi.edu)

Acoustic propagation modeling techniques are often used to estimate the impact of anthropogenic sound sources on the marine environment, and to estimate population density and detection ranges of marine mammals from their vocalizations. Sophisticated propagation models can be used to accurately calculate acoustic transmission loss as a function of range and depth. This is often done along a number of uniformly spaced radials surrounding a sound source and results are interpolated to each simulated animal location. Computational time, detailed input parameters, and interpolation over complex bathymetry can be a hindrance to efficient and accurate results. This work investigates the impact of using simple propagation modeling that avoids interpolation between radials. Differences will be compared between direct and interpolated values as well as between coherent and incoherent transmission loss solutions. The accuracy and efficiency of the different approaches are evaluated by comparing the number of animals that would be taken by the sound field, which are associated with randomized animal locations in Monte Carlo simulations. [Work supported by ONR.]

2:45

1pAB6. Passive acoustic detection and classification of *Ziphius cavirostris*. Ryan Goldhahn (NATO Undersea Research Centre, Viale San Bartolomeo, 400, La Spezia 19126, Italy, goldhahn@nurc.nato.int)

Cuvier's beaked whales, Ziphius cavirostris (Zc), are a species of marine mammals particularly sensitive to anthropogenic noise. Estimating their habitats and abundance is thus of particular importance when planning and conducting active sonar exercises. Since their deep-diving behavior make them difficult to observe visually, passive acoustics is frequently used for detection. A method of automatic detection and classification of Zc is presented based on the inter-click interval, click spectrum, and direction of arrival estimated on a volumetric array. Specifically, click spectra are compared against a signal subspace constructed from eigenvectors of previously identified beaked whale clicks. The direction of arrival is estimated by cross correlating the received click across a three-dimensional array and clicks are classified based on their estimated elevation angle. Additionally, since Zc are known to produce click trains rather than single clicks, detections made without neighbouring detections are discarded as interference. These three criteria are used to detect and classify Zc, in the presence of dolphin clicks and/or other interference. Results are presented on data collected during the SIRENA10 and SIRENA11 experiments conducted by the NATO Undersea Research Centre in the Atlantic Ocean and Mediterranean Sea respectively, and compared against detections made by human operators and a team of visual observers.

3:00

1pAB7. Interesting properties of toothed whale buzz clicks. Odile Gerard (LSM, DGA TN, Avenue de la Tour Royale, Toulon 83000, France, odigea@gmail.com), Craig Carthel, and Stefano Coraluppi (Fusion Technology and Systems, Compunetix, Inc., Monroeville, PA)

Toothed whales are known to click to find prey. The characteristics and repetition rates of the echolocation clicks vary from one species to another, but clicks are fairly regular during the phase in which the animals are searching for prey. Once they have found prey the repetition rate of the clicks increases; these sequences are called buzzes. Some previous work was done to classify Blainville's beaked whale buzz clicks. While we did not succeed to classify these clicks individually because of the variation of their characteristics, we found buzz clicks have slowly varying properties from one click to the next. This similarity permits their association as a sequence using multi-hypothesis tracking algorithms. Thus buzz classification follows the automatic tracking of clicks. We also found that buzz clicks from other toothed whales species often have similar properties. In some cases a variant of this property has been found, whereby sub-sequences of clicks also exhibit slowly varying characteristics.

Session 1pAO

Acoustical Oceanography and Underwater Acoustics: Memorial Session in Honor of Clarence S. Clay II

Dezhang Chu, Cochair
NOAA Fisheries, NWFSC, Seattle, WA 98112

John K. Horne, Cochair
School of Aquatic and Fishery Sciences, University of Washington, Seattle, WA 98195

J. Michael Jech, Cochair
Northeast Fisheries Science Center, Woods Hole, MA 02543

Timothy K. Stanton, Cochair
Woods Hole Oceanographic Institution, Woods Hole, MA 02543-1053

Contributed Papers

1:00

1pAO1. Feature extraction for classification of fish species using the Cepstral analysis. Ikuo Matsuo, Masanori Ito (Department of Information Science, Tohoku Gakuin University, Tenjinzawa 2-1-1, Izumi-ku, Sendai, Miyagi 9813193, Japan, matsuo@cs.tohoku-gakuin.ac.jp), Tomohito Imaizumi, Tomonari Akamatsu (National Research Institute of Fisheries Engineering, Fisheries Research Agency, Hasaki, Ibaraki, Japan), Yong Wang, and Yasushi Nishimori (Furuno Electric Co., Ltd, Nishinomiya, Hyogo, Japan)

Identification and classification of fish species are essential for acoustic surveys of fisheries. The echo from the fish contains components from multiple reflections, including the swimbladder and other organs. The target strength (TS) and temporal structure, which were measured and analyzed by using the broadband signal, were changed dependent on the incident angles and fish species. The cepstral analysis, which was defined as the inverse Fourier transform, was used to discriminate between the spectral pattern associated with the swim bladder and the interference pattern associated with these reflections. Echoes of Japanese jack mackerel (Trachurus japonicus), chub mackerel (Scomber japonicus), and red sea bream (Pagrus major) were measured and analyzed in the sea and tank. It was clarified that the spectral pattern associated the swim bladder was strongly dependent on both the tilt angle and the fish species. [Supported by the Research and Development Program for New Bio-industry Initiatives, and CREST, JST.]

1:15

1pAO2. Exploring the capabilities of an 18-kHz split-beam scientific echosounder for water column mapping and seafloor positioning of methane seeps in the northern Gulf of Mexico. Kevin Jerram, Thomas C. Weber, and Jonathan Beaudoin (Center for Coastal and Ocean Mapping, University of New Hampshire, 24 Colovos Road, Durham, NH 03824, kjerram@ccom.unh.edu)

Underwater methane seeps support diverse biological communities on the seafloor and, in cases of bubble survival to the surface, contribute to the quantity of atmospheric methane. The National Oceanic and Atmospheric Administration (NOAA) ship *Okeanos Explorer* completed two research cruises for seep mapping and characterization in the northern Gulf of Mexico during August and September of 2011 and April of 2012. A 30-kHz Kongsberg EM 302 multibeam echosounder (MBES) and an 18-kHz Simrad EK60 split-beam scientific echosounder were employed to detect and observe seeps during multiple transects over areas of known seep activity at depths of approximately 1500 m. This presentation includes analyses of EK60 data from both research cruises with emphasis on seep mapping in the water column and seep source positioning on the seafloor using EM 302 MBES observations of seeps as benchmarks. Uncertainty associated with interferometric principles employed by the EK60 and limits to midwater positioning capability imposed by its beam pattern are discussed. The importance of sound speed measurement at the transducer face and the effects of refraction correction are estimated by comparison of isovelocity and constant-velocity layer models using sound speed profiles collected during the research cruises.

1:30

1pAO3. Observation of sound fluctuations in presence of internal waves: Difference between refractive and adiabatic regimes. Mohsen Badiey (College of Earth, Ocean, and Environment, University of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu) and Boris G. Katsnelson (Department of Physics, Voronezh State University, Voronezh, Russian Federation)

Angular dependence of sound field on the internal solitary wave propagation in shallow water has been previously shown [JASA, vol.122, p747-760, 2007]. In the presence of moving nonlinear internal waves, the interaction between sound field and internal waves can vary from refractive, to adiabatic, and mode coupling regimes. The mechanism is largely dependent on the angle between the direction of an acoustic track and wave front of internal waves. For small angles, refraction of horizontal field (focusing and de-focusing) can occur. However, for larger angles, the characteristics of the interaction between the acoustic field and internal waves is very different. For example for angle of about 15-20 degrees, the variations of the sound field follow variations of the sound speed profile while for the angle of about 5 degrees, focusing is observed. In this paper experimental observations and the corresponding modeling results showing the difference between the sound field for the small and larger angles are presented. This observation has prompted theoretical investigation [shown in an accompanying paper, Katsnelson, Badiey, et al.] showing the analytical aspect of the problem. Good agreement between the model and the experimental data is shown. [Work was supported by ONR.]

1:45

1pAO4. Mode-2 internal wave generation and propagation-impact on acoustic signal properties. Marshall H. Orr (Acoustics Division, The Naval Research Laboratory, 4555 Overlook Ave SW, Washington, DC 20375, rubyspiral@gmail.com) and Thomas E. Evans (Remote Sensing Division, The Naval Research Laboratory, Washington, DC)

Mode 2 internal wave spatial distributions, generation, propagation and dissipation in the vicinity of bathymetry variably on the continental shelf and near the shelf break of New Jersey, USA will be overviewed for early fall oceanographic conditions. The 3-D spatial and temporal evolution of

the mode 2 waves, as simulated with the Naval Research Laboratory nonhydrostatic Model for Coastal Oceans, will be compared to tow-yo conductivity, temperature and depth (CTD) and high frequency acoustic flow visualization observations of mode -2 internal waves. Sound speed field perturbation and acoustic field spatial and temporal distributions in the presence of mode-2 internal waves will be summarized.

2:00

1pAO5. Ecosystem-based management: What would Clay do? J. Michael Jech (NEFSC, 166 Water St., Woods Hole, MA 02543, michael.jech@noaa.gov)

Fisheries resource management is in a state of transition from managing populations at the species level to managing living marine resources at the ecosystem level. This transition will require changes in the way data are collected, analyzed, integrated, and finally utilized in management decisions. C. S. Clay "Clay" was a pioneer in underwater acoustics, but my first experiences and interactions with him were as a graduate student biologist learning to observe and understand the underwater environment in new and innovative ways. Clay's collaborations with biologists, ecologists, and oceanographers spawned novel methods of integrating and analyzing disparate data sets and many of these methods are being used today. While Clay's influence on fisheries acoustics has been monumental, his approaches to understanding the ocean environment may be most valuable to ecosystem-based management strategies. I will highlight examples of Clay's innovative approaches that have been used and ways they could be applied to ecosystem-based management and research.

MONDAY AFTERNOON, 22 OCTOBER 2012 BENNIE MOTEN A/B, 2:00 P.M. TO 4:30 P.M.

Session 1pID

Interdisciplinary: Introduction to Technical Committee Research and Activities: Especially for Students and First-Time Meeting Attendees

Eric A. Dieckman, Cochair
Dept of Applied Science, College of William and Mary, Williamsburg, VA 23187

Samuel L. Denes, Cochair
Acoustics, Pennsylvania State Univ., University Park, PA 16802

Chair's Introduction—2:00

Invited Papers

2:05

1pID1. Introduction to animal bioacoustics. Holger Klinck (Cooperative Institute for Marine Resources Studies, Oregon State University and NOAA Pacific Marine Environmental Laboratory, Hatfield Marine Science Center, 2030 SE Marine Science Drive, Newport, OR 97365, Holger.Klinck@oregonstate.edu)

Animal bioacoustics (AB) covers all matters related to the production, transmission, and reception of sound in nature, as well as the investigation and use of natural sound by people and impacts of anthropogenic sounds by on animals. Topics include animal communication; sound production mechanisms; sound reception mechanisms; evolution of sound production and hearing mechanisms; effects of acoustic propagation on natural sounds; sound detection, classification, localization and tracking; estimating populations and population density; impact of human-generated noise on animals; and a variety of other topics. All animals, and indeed all organisms, are considered within the scope of AB, though the most common taxa are marine mammals, birds, primates and other mammals, fishes, frogs and other amphibians, and insects.

2:15

1pID2. An introduction to the Biomedical Acoustics Technical Committee. Robert McGough (Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824, mcgough@egr.msu.edu)

An overview of the activities of the Biomedical Acoustics Technical Committee will be presented, along with a brief survey of some of the ongoing research projects in Biomedical Acoustics. For example, the Biomedical Acoustics Technical Committee organizes a student poster competition every year, and special sessions are arranged at each meeting to highlight some of the latest research in the diagnostic and therapeutic applications of acoustics. Some of the topics include ultrasound imaging, high intensity focused ultrasound, ultrasound mediated drug delivery, ultrasound contrast agents, ultrasound-induced cavitation, and ultrasound elastography, among several others. In this presentation, selected examples of active research topics in Biomedical Acoustics will also be described in greater detail.

2:25

1pID3. Psychological and physiological acoustics: From sound to neurons to perception … to clinical and engineering applications. Michael Heinz (Speech, Language, and Hearing Sciences & Biomedical Engineering, Purdue University, 500 Oval Drive, West Lafayette, IN 47907, mheinz@purdue.edu)

Psychological and physiological acoustics includes a wide range of multidisciplinary approaches and topics, ranging from basic science to clinical and engineering applications. Research in this area of acoustics is concerned with questions such as how sound is processed once it enters the auditory system (both in humans and other species), how sound is used by organisms to help them communicate and navigate their environment, and how experience and pathology can alter the signal processing of sounds through neural plasticity. Topics include the biomechanics of the middle and inner ear; the neuroscience of the auditory nerve, brainstem, and cortex (including both upward and downward connections); and behavioral and cognitive studies of auditory perception. This talk provides a brief overview of several current areas of research, ranging from basic questions about the neural codes that represent various features of sound in the peripheral and central auditory systems, to clinical applications including the development and improvement of cochlear, brainstem, and midbrain implants that bypass the peripheral auditory system to restore hearing to people with profound hearing loss.

2:35

1pID4. An introduction to current research in Speech Communication. Tessa Bent (Department of Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

Speech is a highly complex acoustic signal yet most people can produce and understand speech rapidly and without error. Researchers in Speech Communication—including speech and hearing scientists, linguists, psychologists, and engineers—are deepening our understanding of this amazing process. Topics covered by the Speech Communication Technical Committee include speech production, perception, and acquisition; acoustic phonetics; speech and hearing disorders; neuroscience of speech production and perception; speech intelligibility; communicative aspects of singing; speaker classification and identity; audiovisual speech perception; and speech processing and technology. This presentation will provide a brief overview of some current research areas within Speech Communication including speech acquisition in first language, second language, and bilingual contexts; the impact of speech and hearing disorders on communication; imaging and modeling of speech production using new technologies; and adaptation to novel speech stimuli throughout the lifespan.

2:45

1pID5. Architectural acoustics—Space for sound, and you. Alex Case (Sound Recording Technology, University of Massachusetts Lowell, 35 Wilder St, Suite 3, Lowell, MA 01854, alex@fermata.biz)

The discipline of Architectural Acoustics consistently produces more than 100 papers across 6 or more special sessions, at each meeting of the ASA. Student paper awards, student design competitions, and Knudsen lectures augment these activities. Joint sessions, particularly with Noise, Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics, add more still to the architectural acoustics goings-on at every ASA conference. The sphere of influence is not limited to ASA alone, as TCAA members participate in the Green Construction Code of the International Code Council, Society of Motion Picture and Television Engineers Study Group: Movie Theater Sound System Measurement and Adjustment Techniques, Classroom Acoustics Standards, the American Institute of Architects Continuing Education System, and more. This busy committee also produces a steady stream of publications documenting recent work and deciphering standards for key stakeholders. Anyone with an interest in the field will find many opportunities to advance their own expertise, build a network of colleagues, friends and mentors, and contribute to the essential activities of the Technical Committee on Architectural Acoustics.

2:55

1pID6. Introduction to the Structural Acoustics and Vibration Technical Committee. James E. Phillips (Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608, jphillips@wiai.com)

The Structural Acoustics & Vibration Technical Committee (SAVTC) includes the study of motions and interactions of mechanical systems with their environments and the methods of their measurement, analysis, and control. This talk will provide a broad overview of the many research areas of interest to SAVTC. A few topics will be explored in more depth to provide background on some of the more common analysis methods used by members of the technical committee.

3:05–3:20 Break

3:20

1pID7. Noise and its impact on our world. Erica Ryherd (Mechanical Engineering, Georgia Institute of Technology, Mechanical Engineering, Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu)

Noise invades all aspects of our lives. The word noise is actually derived from the Latin word “nausea”, with one possible connection being that unpleasant sounds were made by seasick passengers or sailors in ancient times. In modern times, the demand for noise research and consulting has intensified in concert with rising population densities, growing industrialized societies, escalating demands from consumers, and increasingly common standards and legislation related to noise. The Acoustical Society of America Technical Committee on Noise (TC Noise) is concerned with all aspects of noise, ranging from noise generation and propagation, to active and passive methods of controlling noise, to the effects of noise on humans and animals. This talk will explore the broad topic of noise and its impact on our world.

3:30

1pID8. The Engineering Acoustics Technical Committee. Michael V. Scanlon (US Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

Engineering Acoustics Technical Committee (EATC) encompasses the theory and practice of creating tools for investigating acoustical phenomena and applying knowledge of acoustics to practical utility. This includes the design and modeling of acoustical and vibrational transducers, arrays, and transduction systems in all media and frequency ranges; instrumentation, metrology, and calibration; measurement and computational techniques as they relate to acoustical phenomena and their utility; and the engineering of materials and devices. EATC's recently sponsored special sessions attracted members from all areas and is an enabler for overlapping interests: single crystal piezoelectrics, transducer design, vector sensors, materials for high power sonar, micromachined silicon mics, multimedia, 3-D audio, pro-audio, parametric sources, speakers, transduction, projection, materials processing and manufacturing, piezocomposites, sensor fusion, signal processing hardware, data communication, condition-based monitoring of machinery and processes, solid-state sensors and actuators, hearing aids and ear impedance, thermoacoustic refrigerators and engines, wind turbines, energy harvesting, computer modeling methods, acoustic barriers & noise control, array applications, metamaterials, systems for underwater vehicles, mufflers & silencers, borehole acoustics, flow noise mitigation methods, and non-destructive testing with ultrasound & non-contact methods.

3:40

1pID9. Signal processing in acoustics: Why not get involved? R. L. Culver (ARL, Penn State University, PO Box 30, State College, PA 16804, rlc5@psu.edu)

The Signal Processing Technical Committee (SPTC) of the ASA provides a forum for discussion of signal processing techniques that transcend any one application. Signal processing research typically presented at ASA meetings includes techniques that show promise in one application - say underwater acoustics - but may also have application to other areas, for example, speech processing or room acoustics. There are two good reasons to get involved in the SP TC. First, since signal processing is an important aspect of many acoustic research areas, you will have the opportunity to better understand new and potentially useful tools. Second, we are a small TC and you can make an immediate contribution.

3:50

1pID10. An introduction to the Physical Acoustics Technical Committee activities. Steven L. Garrett (Grad. Prog. in Acoustics, Applied Research Lab, Penn State University, P.O. Box 30, State College, PA 16804, sxg185@psu.edu)

The primary activity of any ASA Technical Committee is to use collective wisdom of the Committee's membership to determine which research topics within its specialization area are most active and interesting. Based on that assessment, the Committee organizes special sessions at future meetings that will bring together experts from those areas, not necessarily limited to the Society members, who can share interesting results and provide guidance regarding the directions that will lead to further understanding. In Physical Acoustics, that is a particularly daunting challenge given the scope of topics that fall within its purview: use of sound to probe material properties, sound propagation and attenuation mechanisms on this planet and in other parts of the universe, and physical effects of sound and its interaction with other forms of radiation, all of which could also go well beyond the limitations of a linear acoustical theory. Needless to say, involvement in debates about "what's hot" is both interesting and educational. Other activities include proposals for Technical Initiatives that allocate ASA resources. Recently, PATC received funding to sponsor a demonstration session at the Physical Acoustics Summer School.

4:00

1pID11. Overview of Acoustical Oceanography Technical Committee. Aaron Thode (SIO, UCSD, 9500 Gilman Dr, MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu)

The Acoustical Oceanography (AO) technical committee focuses on the use of sound to study physical and biological processes in the ocean. The broad scope of the committee ensures that many of the research topics overlap those in Underwater Acoustics and Animal Bioacoustics. This presentation will review representative aspects of AO, including long range acoustic tomography, air-sea interactions via studying bubble plumes, fisheries acoustics, and marine mammal acoustic tracking.

4:10

1pID12. An introduction to the Underwater Acoustics Technical Committee. Marcia J. Isakson (Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78713, misakson@arlut.utexas.edu)

Acoustics is considered the best means of remote sensing in oceans, lakes, and estuaries due to the high attenuation of electromagnetic radiation in water. The members of the underwater acoustics technical committee are concerned with the generation and propagation of sound in an underwater environment as well as acoustic reflection and scattering from the seabed, sea surface and objects in the water column and on or beneath the seabed. In this talk, a short history of underwater acoustics will be followed by an overview of the current state of research in the field.

4:20

1pID13. An overview of the Musical Acoustics Technical Committee. Andrew Morrison (Joliet Junior College, 1215 Houbolt Rd, Natural Science Department, Joliet, IL 60431, amorrison@jjc.edu)

The technical committee on musical acoustics (TCMU) is concerned with the application of science and technology to the field of music. Many of the sessions organized by the TCMU focus on a particular family of musical instruments. At many Acoustical Society meetings the TCMU has arranged for musical performances related to one or more special sessions. The TCMU has also arranged for tours of sites in the community around past meeting locations. An overview of what students and relative newcomers to the society can expect from the TCMU in general will be presented as well as highlights of what to see at this meeting.

Session 1pMU

Musical Acoustics: Acoustics of the Pipe Organ

Uwe J. Hansen, Chair
Indiana State University, Terre Haute, IN 47803-2374

Invited Papers

1:00

1pMU1. The Research Organ at the Fraunhofer IBP in Stuttgart. Judit Angster (Acoustics, Fraunhofer IBP, Nobelstr 12, Stuttgart 70569, Germany, angster@ibp.fhg.de) and András Miklós (Applied Acoustics, Steinbeis Transfer Center, Stuttgart, Baden Wuerttemberg, Germany)

At the Fraunhofer IBP in Stuttgart a research organ has been built for scientists. Its transparency and unique design allows the demonstration of research results, the investigation of technical and sound problems in organ building as well as the audible testing of sound ideas. Some of the special design features are: The wind system can be switched from traditional to innovative design. Newly developed swell shutters which allow a better dynamic of the sound are mounted in the swell organ. The dimensioning of pipes and wind systems was conducted by means of scaling software developed within the context of European research projects. One wind chest of a division can be exchanged to allow the testing of valves and pipe layouts. In order to test newly designed stops a blind slider is available. There are several blind grooves to analyze the effect of wind flow, of resonances in the grooves and of different outlet holes on the pipe sound. The blowers are driven by a frequency converter for continuous adjustment of wind pressures. The motion of a beating and a free reed can be visualized by means of a stroboscope installed. Some of the research results will be demonstrated.

1:20

1pMU2. The challenge of orchestrating for the organ and the orchestra. Jason L. Raymond (Biomedical Engineering Program, College of Engineering and Applied Science, University of Cincinnati, CVC 3940, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, raymonjl@mail.uc.edu), Christina Haan (College-Conservatory of Music, University of Cincinnati, Cincinnati, OH), and Christy K. Holland (Department of Internal Medicine, Division of Cardiovascular Diseases, College of Medicine, University of Cincinnati, Cincinnati, OH)

Hector Berlioz (1803-1869), though not an organist himself, included descriptions of organ stops, registration, pitch range, and use in orchestral compositions in his Grand Traité d'Instrumentation et d'Orchestration modernes (1855). When discussing the unsuitability of combining the organ with the orchestra, he stated, "…the even and uniform tones of the organ can never fuse completely with the extremely variable sounds of the orchestra…." The objective of our study was to use this statement as a hypothesis to be tested. The relative dissonance of selected organ stops in combination with four orchestral instruments was analyzed using quantitative frequency analysis techniques. Three organs were chosen for study, each built by a different organ company, and each in a different acoustic setting. Four orchestral instruments (violin, flute, oboe, and horn) were chosen for their contrast in timbre and method of sound production. The differences in pitch between the overtones of the organ and instrument were compared in order to quantify the relative dissonance produced by each combination as a function of critical bandwidth of the human ear. In general, the combinations with the oboe and flute resulted in frequency differences less than 2% of the critical band for each overtone, which were perceived as minimally dissonant. In some cases, the violin and horn overtones exceeded 2% and 4% of the critical band, respectively.

1:40

1pMU3. Scanning vibrometry studies of reed dynamics in reed organ pipes. Thomas M. Huber, Lucas Seewald (Physics, Gustavus Adolphus College, 800 W College Ave, Saint Peter, MN 56082, huber@gac.edu), and Charles Hendrickson (Hendrickson Organ Company, Saint Peter, MN)

Unlike most other wind instruments, in reed organ pipes it is not the length of the resonator tube that primarily determines the pitch, but the length of the vibrating reed. The reed, its fluid-structure interaction with the airflow, and feedback from the resonator form a very complicated dynamical system. By utilizing a Polytec PSV-400 scanning laser Doppler vibrometer, it is possible to monitor the operating deflection shapes of a vibrating reed, while using a microphone to monitor the sound production. In the current study, an experienced organ builder made adjustments to the reed/resonator/airflow parameters, both within a musically appropriate range, and for ranges outside of the norm. Results will be presented of the acoustical and vibrational spectral envelope, and the corresponding deflection shapes for each harmonic, for a range of different adjustments to the system. The vibrational results for the reed will be compared to measurements where the non-contact ultrasound radiation force is used to excite the natural resonance frequencies and deflection shapes of the reed in the absence of airflow. In addition to steady-state deflection shapes, results will be presented for the transient dynamics of the system as the airflow into the pipe is initiated.

2:00

1pMU4. Categorical sound characteristics of free-reed pipe-organ stops. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180, braasj@rpi.edu)

Organ pipes with free reeds became very popular in parts of Europe at the end of the 18th century, but vanished about 100 years later following a critical discussion about their characteristic sound qualities. The research reported here is based on the measurement and analysis of 11 free-reed organ stops and 15 striking-reed organ stops, as well as 7 flue-pipe organ stops for comparison purposes. The results show that the sound characteristics between free-reed and striking-reed pipes are perceptually distinguishable. Both types of reed stops have their unique set of characteristics. In general, striking-reed pipes have a higher spectral centroid than free-reed pipes. While the overtone spectra of the latter can be fairly similar to those of flue pipes, their onset characteristics do not match the other two types of pipe organ stops. Their fundamental frequencies gradually increase by a semitone over the duration of approximately 60 ms. The frequency shift can be larger for striking-reed pipes, but the duration of the shift is less than half the value for free reeds. In contrast, flue pipes do not usually show such a characteristic frequency shift.

2:20

1pMU5. Aeolian-Skinner Opus 1309 in the Community of Christ Auditorium, Independence, Missouri, USA. Michael Quimby (Quimby Pipe Organs, Inc., PO Box 434, 208 Marshall Street, Warrensburg, MO 64093, qpo1@earthlink.net)

The 113-rank Auditorium organ located in Independence, Missouri, was built by the Aeolian-Skinner Organ Company of Boston, Mass. Immediately after its installation in 1959, the organ became - and remains - perhaps the most important example of the company's work from the period. The commanding display of exposed Great, Positiv, and Pedal pipework forms the visual centerpiece of the massive conference chamber which seats nearly 5,800 people. The main organ is framed by nineteen acoustical clouds suspended above and in front of it, and by choir seating and the large rostrum beneath it. The entire room is covered by a huge dome, culminating in an oculus rising some 100 feet above the floor. G. Donald Harrison, President and Tonal Director of Aeolian-Skinner and one of the twentieth century's most influential organ builders, was responsible for the organ's initial design and specification in the mid-1950s. Following Mr. Harrison's untimely death in 1956, Joseph Whiteford was appointed Tonal Director and, in collaboration with consultants Catharine Crozier and Harold Gleason, finalized the design and formulated the organ's pipe scales. The organ is a superb example of the "American Classic Organ," a concept and design developed by Aeolian-Skinner. (Text by Thomas Brown)

2:40

1pMU6. Acoustics of the pipe organ. Jan Kraybill (Community of Christ Headquarters, 1001 W Walnut St, Independence, KS 64050, jkraybill@CofChrist.org)

An organist's perspective on pipe organ acoustics. The range of frequencies produced by a large pipe organ covers 10 Hz to over 10,000 Hz. Three significant pipe organs in the Kansas City metro area will be discussed: the Julia Kauffman Casavant at the Kauffman Center for the Performing Arts in Kansas City (installation completed early 2012) and the two magnificent instruments at Community of Christ Headquarters in Independence, MO (installations in 1993 and 1959). The acoustic qualities of individual pipes, groups of pipes, and the rooms in which these instruments reside affect the approach which these organs' builders took in designing these one-of-a-kind pipe organs, and the approach each artist takes when making music with them.

Session 1pNS

Noise, Physical Acoustics, Animal Bioacoustics, and ASA Committee on Standards: Outdoor Sound Propagation

Siu-Kit Lau, Cochair
University of Nebraska-Lincoln, Omaha, NE 68182-0816

Kai Ming Li, Cochair
Mechanical Engineering, Purdue University, West Lafayette, IN 47907-2031

Invited Papers

1:00

1pNS1. Excess attenuation and effective impedance associated with rough hard ground. Keith Attenborough, Imran Bashir, and Shahram Taherzadeh (Design, Development, Environment and Materials, The Open University, Walton Hall, Milton Keynes, Buckinghamshire MK7 6AA, United Kingdom, k.attenborough@open.ac.uk)

Although ground effect has been studied widely and is an accepted component of outdoor sound prediction schemes exploitation of ground effect has been restricted to the development of porous road surfaces which influence traffic noise generation as well as propagation. Relatively little attention has been paid to the potential for exploiting the effectively finite impedance associated with roughness small compared with the incident wavelengths on an otherwise acoustically-hard ground or to the potential usefulness of the wide variation in soft ground effects that are available. Results of recent laboratory and field measurements on artificially rough hard surfaces including the comparative acoustical performances of randomly- and periodically-spaced roughness are presented. The laboratory data have been used to study the influences of roughness shape and spacing and surface wave formation. Practical outdoor realisations of artificially roughened hard surfaces have utilised low brick wall configurations and their relative merits have been studied through loudspeaker and car pass by measurements and Boundary Element calculations.

1:20

1pNS2. Estimation of blast noise sound fields over large regions using noise monitors and geostatistical models. Edward T. Nykaza, Michael J. White (US Army Corps of Engineers, Engineer Research and Development Center, 2902 Newmark Drive, Champaign, IL 61822, edward.t.nykaza@erdc.dren.mil), D. Keith Wilson (US Army Corps of Engineers, Engineer Research and Development Center, Hanover, NH), and Anthony A. Atchley (College of Engineering, The Pennsylvania State University, University Park, PA)

This study explores the feasibility of accurately estimating blast noise levels over a large region between and beyond noise monitoring stations using geostatistical models. The potential improvements over propagation modeling include faster computations, fewer assumptions, and improved accuracy. The estimation models explored include kriging, simple interpolation, and models that include meteorological and terrain parameters commonly incorporated into outdoor sound propagation models. The estimation models are evaluated using both experimentally measured and simulated noise monitor data gathered under various atmospheric conditions in several large regions (e.g., greater than 16 km^2). The performance of using geostatistical-based estimation models is discussed in terms of the uncertainty of sound pressure field estimates, sensitivity to atmospheric variability, sensor density and geometry configuration, and model validation.

1:40

1pNS3. Aircraft sound transmission in homes categorized by U.S. climate regions. Erica Ryherd and Nathan Firesheets (Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu)

Current aircraft noise guidelines are based primarily on outdoor sound levels. However, human perception is highly related to indoor response, particularly for residences. A research project is being conducted that provides insight into how typical residential dwelling envelopes affect indoor sound levels. A focus is being placed on non-sonic boom aircraft noise, using continuous noise signatures of commercial aircraft overflights. Typical construction types in various U.S. climate regions have been identified and used to develop model predictions of indoor noise levels. Further, the impacts of systematically altering construction variables such as construction material and window to wall ratio is being investigated. Results will be used to understand trends for expected noise reduction for typical construction types around the U.S.

2:00

1pNS4. Recent advances in sound propagation above a non-locally reacting ground. Kai Ming Li (Mechanical Engineering, Purdue University, 140 South Martin Jischke Drive, West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

In the absence of atmospheric effects, the sound fields above a locally reacting ground can be accurately predicted by the Weyl-van der Pol formula. The solution is based on an asymptotic analysis to yield two terms: a direct and a ground reflected wave terms. The reflected wave term can be written as a product of a spherical wave reflection coefficient and the sound reflected from a rigid ground. In the contrary, it is more challenging to derive a similar formula for the sound fields above non-locally reacting grounds. In the past, an approximation in the same form as the Weyl-van der Pol formula has been used which becomes inadequate for layered grounds. In this presentation, a brief review of the asymptotic analysis will be discussed. An overview of the analytical and numerical approaches will be presented for obtaining accurate prediction of sound fields above the non-locally reacting ground. It will be further demonstrated that the reflection coefficient can be split exactly into two terms - a plane wave reflection coefficient and a ground wave term involving the boundary loss factor. The correlation between the numerical distance and the location of the surface wave pole will be examined.

2:20

1pNS5. A hybrid computational method for computing urban canyon sound fields exhibiting roughness. Carl Hart and Siu-Kit Lau (University of Nebraska - Lincoln, 1110 S 67th St, Omaha, NE 68182, carl.hart@huskers.unl.edu)

Predicting the three-dimensional sound field within an urban canyon is essential for urban noise assessment. Several analytical and computational methods exist to predict the canyon sound field. Generally building facades can be characterized as both geometrically reflecting and dispersive. The image source method is well suited to scenarios where surfaces are geometrically reflecting. Given dispersive surfaces the radiosity method predicts the sound field well. On the other hand a canyon exhibiting geometric and dispersive facades requires a computationally intensive technique such as the finite-element method (FEM) or the boundary-element method (BEM). Due to computational limitations neither the FEM nor BEM are well suited for computing large, unbounded, three-dimensional, urban canyon domains. A prediction method which synthesizes adaptive beam tracing and a secondary model for edge diffraction serves as an alternative technique for analyzing urban canyon acoustics, which contain façades exhibiting both geometric and dispersive surfaces. Advantages of the hybrid method include the ability to model unbounded domains simply, no requirement for discretizing geometric boundaries, and the ability to model sound propagation in the time domain. The magnitude of acoustic dispersion from a surface is related to surface roughness. The effect of surface roughness on the canyon sound field is investigated.

Contributed Papers

2:40

1pNS6. A single-bounce method for estimating impulse propagation and attenuation in a forest. Michelle E. Swearingen (US Army ERDC, Construction Engineering Research Laborartory, P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil) and Donald G. Albert (US Army ERDC, Hanover, NH)

There are numerous methods for determining the impact of interactions with multiple scattering objects on a propagating signal. Many of these methods assume that the signal is continuous in time, allowing one to neglect time dependence. Additionally, these methods make assumptions about the spatial distribution of scattering objects, such as lattice or random, and many assume that the scattering objects are all identical. Real forests often have trees of varying trunk diameters and may be arranged in a grid, in clusters, or random, and the assumptions of uniform size and particular spatial distributions introduce error into the assessments. A simple model, based on single bounces from trunks, is developed to begin estimating the propagation of an impulsive signal through this complicated multiple scattering environment. The scattering algorithm takes the frequency-dependent radial scattering pattern of a cylinder into account. Results are compared to data collected in a forest stand where the locations and diameters of trunks were carefully recorded. These comparisons provide insight into whether a single, discrete bounce method is sufficient for modeling impulse propagation in this complex environment, or whether multiple discrete bounces or some other method should be explored.

2:55

1pNS7. Investigations of measured temperature and wind effects on outdoor sound propagation. Lauren M. Ronsse, Dan Valente, Edward T. Nykaza, and Michael J. White (Construction Engineering Research Laboratory, US Army Engineer Research and Development Center, P.O. Box 9005, Champaign, IL 61826-9005, Lauren.Ronsse@erdc.dren.mil)

Temperature and wind effects on outdoor sound propagation have been well-studied in numerous theoretical investigations, leading to a number of commonly held beliefs about how weather affects outdoor propagation. For example, louder sound levels are expected when downwind propagation and temperature inversions are present, whereas lower levels are expected when upwind propagation and temperature lapses exist. However, the validity of such relationships has not been rigorously tested in the field for various terrains at long distances from an outdoor source. One is justified in questioning the validity of these relationships due to the difficulty of adequately modeling the dynamic atmosphere along the entire propagation path. This study examines some commonly held notions of outdoor sound propagation by experimentally investigating the effects of temperature gradients and wind speed/direction on sound propagation at long distances from a typical impulsive source. Temperature and wind conditions measured near the source and along the line of sound propagation are correlated with the received sound pressure levels recorded at distances up to 15 km from the source. The variability of peak sound pressure levels occurring under similar wind and temperature conditions is assessed, demonstrating that much complexity underlies these common aphorisms.

3:10–3:30 Break

3:30

1pNS8. A modified saddle point method for predicting sound penetration into a rigid-porous ground. Hongdan Tao and Kai M. Li (Purdue University, 140 S. Martin Jischke Drive, West Lafayette, IN 47907, htao@purdue.edu)

An approximate analytical formula has been derived for the prediction of sound penetration into a semi-infinite rigid porous ground due to a monopole source. The sound fields can be expressed in an integral form that is amenable to analytical and numerical analyses. A modified saddle point method is applied to evaluate the integral asymptotically that leads to a closed form expression. The validity of the asymptotic formula is confirmed by comparing with the numerical results computed by the fast field formulation and the direct evaluation of the integral. It has been demonstrated that the present analytical formula is sufficiently accurate to predict the penetration of sound into the semi-infinite rigid porous ground. Furthermore, the sound fields predicted by modified saddle point are compared with that computed by the double saddle point method. It is found that these two asymptotic schemes give precise solutions at far fields but the modified saddle method is more accurate at the near field especially when both the source and receiver are close to the air/ground interface. [Research is partially funded by the China Research Scholarship Council.]

3:45

1pNS9. Wind noise reduction in a non-porous subsurface windscreen. Allan J. Zuckerwar (Analytical Services and Materials, 1052 Research Drive, Hampton, VA 23666-1340, ajzuckerwar@yahoo.com), Qamar A. Shams, and Keith Knight (NASA Langley Research Center, Hampton, VA)

Measurements of wind noise reduction were conducted on a box-shaped, subsurface windscreen made of closed cell polyurethane foam. The windscreen was installed in the ground with the lid flush with the ground surface. The wind was generated by means of a fan, situated on the ground, and the wind speed was measured at the center of the windscreen lid with an ultrasonic anemometer. The wind speed was controlled by moving the fan to selected distances from the windscreen. The wind noise was measured on a PCB Piezotronics 3" electret microphone. Wind noise spectra were measured with the microphone exposed directly to the wind (atop the windscreen lid) and with the microphone installed inside the windscreen. The difference between the two spectra comprises the wind noise reduction. At wind speeds of 3, 5, and 7 m/s, the wind noise reduction is typically 15 dB over the frequency range 0.1-20 Hz.

4:00

1pNS10. Measurements of acoustic transmission loss over a rough water surface. Cristina Tollefsen and Sean Pecknold (Defence Research and Development Canada - Atlantic, P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, cristina.tollefsen@gmail.com)

Recent interest in understanding acoustic propagation over rough water surfaces has been driven largely by the increasing presence of offshore wind turbines and concerns about the potential for community noise disturbance. In addition, there has been interest in evaluating directional acoustic hailing devices for use at sea, in determining potential environmental impact of naval gunfire exercises, and in understanding the in-air acoustic footprint of maritime-based military assets. Measurements of acoustic transmission loss over a rough water surface were made during the period of 3 Oct and 7 Dec 2011 near Halifax, Nova Scotia, Canada. The acoustic source was a propane cannon, firing four consecutive shots once per hour during daylight hours. A receiver was positioned at ranges of 2 km to 7.5 km from the source, with a clear line-of-sight across the water, for periods ranging from 5-21 days at each location. Temperature, wind velocity, relative humidity, and ocean surface wave height were acquired from a variety of sources, including point measurements, radiosonde profiles, and output from an atmospheric forecasting model (Environment Canada's Global Environmental Multiscale [GEM] model). The measured values of transmission loss are compared to results obtained with a parabolic equation-based atmospheric acoustic propagation model.

4:15

1pNS11. Noise exposure profiles for small-caliber firearms from 1.5 to 6 meters. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), Gregory A. Flamme (College of Health and Human Services, Western Michigan University, Kalamazoo, MI), Edward L. Zechmann, Caroline Dektas (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, Cincinnati, OH), Deanna K. Meinke (Audiology and Speech-Language Sciences, University of Northern Colorado, Greeley, CO), Michael Stewart (Communication Disorders Department, Central Michigan University, Mount Pleasant, MI), James E. Lankford (Allied Health and Communication Disorders, Northern Illinois University, DeKalb, IL), and Donald S. Finan (Audiology and Speech-Language Sciences, University of Northern Colorado, Greeley, CO)

Small caliber firearms (rifles, pistols and shotguns) are commonly used at outdoor firing ranges for training in shooting skills, job qualification and for recreation. Firearm noise from fifty-four weapons was measured at an outdoor range in the near field (6 meters and closer) of the weapons using a radial array of 18 microphones centered on the shooter's head. Each weapon was fired five times and the microphone array was sampled at 200 kHz with at least 16-bit resolution. Peak sound pressure levels and damage risk criteria (e.g. MIL-STD 1474D, 8-hour Equivalent A-weighted Level (LAeq8), and Auditory Hazard Assessment Algorithm for Humans (AHAAH)) were computed for each microphone and compared across weapon type, caliber and load. The acoustic propagation from the muzzle to the microphone was modeled using a simple image source over a reflecting plane. The impedance of the ground was estimated from the observed data and was used to compare the measured waveforms with the estimated waveforms. These data will be used to model the exposures for multiple shooters and observers standing at or behind the firing line.

4:30

1pNS12. Effects of inclusion shapes within rigid porous materials on acoustic performance. Hyun Hong and Siu-Kit Lau (Charles W. Durham School of Architectural Engineering and Construction, University of Nebraska-Lincoln, 1110 S. 67th Street, Omaha, NE 68182, slau3@unl.edu)

The present study investigates the influence of various shapes of inclusions having same volume embedded in a porous rigid material. Previous studies showed improvement of the broadband sound absorption with particular shapes of inclusions. However, different volumes of the inclusions have been considered; therefore, the bulk densities are not the same for comparison. The present study extends the investigations of inclusions in porous materials with same volume (or bulk density) to eliminate the influence by the change of bulk density. The effects of shape will be discussed. Finite element modeling will be used for this study. Total four different shapes: circle, square, ellipse, and triangle, have been studied at various orientations. It has been found that specific configurations can be able to improve the broadband sound absorption compared with reference (no inclusion). It is being expected that a better control of sound absorption of porous materials at desired frequency range can be achieved with the results of the present study.

Session 1pPA

Physical Acoustics and Biomedical Acoustics: Memorial Session for Wesley Nyborg

Lawrence A. Crum, Cochair
Applied Physics Laboratory, University of Washington, Seattle, WA 98105

Junru Wu, Cochair
Physics, University of Vermont, Burlington, VT 05405

Chair's Introduction—1:00

Invited Paper

1:05

1pPA1. Wesley Nyborg and bioacoustics at the University of Vermont. Junru Wu (Physics, University of Vermont, Cook Bldg, Burlington, VT 05405, jwu@uvm.edu)

Wesley Nyborg, Physics Professor Emeritus at the University of Vermont, passed away on September 24, 2011 after a full and wonderful life of 94 years. Wes came to the University of Vermont in 1960 where he did his most pioneering research in microstreaming, acoustic radiation pressure and bioeffects of ultrasound. He was considered as one of most influential pioneers by the international biomedical ultrasound community. Wesley Nyborg has developed an active bioacoustics and biomedical ultrasound research in the university. His research was continuously supported over 20 years by NIH. In this presentation, his research and life with the university will be reviewed.

Contributed Papers

1:25

1pPA2. Wes and Ed and the minus sign—A tale of two giants. Charles C. Church (NCPA, University of Mississippi, 1 Coliseum Drive, University, MS 38677, cchurch@olemiss.edu)

Once upon a time there was a young lad who liked to stroll through the Land of Bioacoustics. It was a pleasant land filled with many things that this fellow found very interesting indeed, but there were also giants. These giants had powerful names like Wes and Ed, and they understood many things that were hidden from the lad. One of these was how to calculate the force exerted by a traveling wave on an object in an acoustic field. With some trepidation, the fellow asked the giants to explain the proper way to make the calculation. "Certainly lad," said Wes, "Here's how to do it." "Oh," said Ed, "but what about this?" Each gave him a paper, and he studied them hard, turning them this way and that, and what did he find? The answers were the same. Well almost. One had a minus sign where the other did not. "I don't understand," said the lad, "How can this be?" "We're really not sure," said the giants, "We'll have to discuss it. We'll tell you tomorrow, or in a month or a year." To learn what they said finally, come to Kansas City to hear.

1:40

1pPA3. Acoustic and optical characterization of ultrasound contrast agents via flow cytometry. Camilo Perez, Andrew Brayman (Center for Industrial and Medical Ultrasound (CIMU), University of Washington Applied Physics Laboratory, 1013 NE 40th Street, Seattle, WA 98105-6698, camipiri@uw.edu), Juan Tu (Key Laboratory of Modern Acoustics, Nanjing University, Nanjing, Jiang Su, China), Jarred Swalwell, Hong Chen, and Tom Matula (Center for Industrial and Medical Ultrasound (CIMU), University of Washington Applied Physics Laboratory, Seattle, WA)

Characterizing ultrasound contrast agents (UCAs) involve measuring the size and population distribution. However, these instruments do not allow for characterization of shell properties, which are important for (1) stability to administration and circulation throughout the vasculature; (2) UCA response to ultrasound; and (3) conjugating ligands for molecular imaging. Thus it is critical to understand the physical and rheological properties of shells. We previously developed a light scattering technique to characterize the shell properties of UCAs [Guan and Matula, JASA, Vol 116(5), 2004; Tu, et al., IEEE Trans. Ultrason., Ferroelec., and Freq. Control, vol. 58(5), 2011]. The most recent manifestation involves a flow cytometer modified with a custom square quartz flow cell in place of the standard nozzle and fluid jet. Acoustic coupling to the carrier sheath fluid and UCA samples occurred through a PZT bonded to one side of the flow cell. The PZT-driven UCA oscillations were processed and fitted to the Marmottant UCA model. Shell properties for UCAs (including Definity, Optison, SonoVue, and even homemade bubbles) were determined. The focus of this talk will be on pressure calibration and additional measurements of unpublished data from Optison and homemade bubbles. (Funded in part by the Life Sciences Discovery Fund #3292512)

1:55

1pPA4. Gauging the likelihood of stable cavitation from ultrasound contrast agents. Kenneth B. Bader and Christy K. Holland (Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Kenneth.Bader@uc.edu)

Clinical ultrasound scanners use the Mechanical Index (MI) to gauge the potential for bioeffects due to inertial cavitation. However, the advent of ultrasound contrast agents (UCA) introduces nucleation mechanisms for bubble activity far different than that assumed in the development of the MI. Such exogenous agents promote bubble activity at substantially lower acoustic pressures than those required for inertial cavitation. The onset of this type of gentle bubble activity is within a stable cavitation regime. The minimum thresholds of both stable cavitation (as indicated by the onset of subharmonic oscillations) and the rupture of UCA were numerically calculated as a function of frequency. Both of these thresholds were found to depend linearly on frequency, and a "cavitation index" will be introduced. This index will be compared to the MI and compared to bioeffects studies in the literature. This cavitation index is not intended to replace the MI. Rather, it may be used to gauge the destruction of UCA, or promote bubble activity

to induce beneficial bioeffects mediated by stable cavitation. This work was supported by the National Institutes of Health, grant numbers NIH RO1 NS047603, NIH RO1 HL059586, and NIH RO1 HL74002.

2:10

1pPA5. Engineering tissues with ultrasound heating, radiation force, and cavitation. Diane Dalecki (Biomedical Engineering, University of Rochester, 310 Goergen Hall, P.O. Box 270168, Rochester, NY 14627, dalecki@bme.rochester.edu) and Denise C. Hocking (Pharmacology and Physiology, University of Rochester, Rochester, NY)

Wesley Nyborg was a pioneer in the field of biomedical ultrasound. His theoretical and experimental work forms the foundation for our understanding of the biological effects of ultrasound. He developed fundamental theories of the physical mechanisms of interaction of ultrasound with tissues, including heating, radiation force, and cavitation. In this presentation, we discuss our development of three ultrasound technologies that employ heating, radiation force, or cavitation to address important challenges in tissue engineering. One technology uses radiation forces developed in an ultrasound standing wave field to spatially pattern cells within engineered tissues. The resultant ultrasound-induced patterning can lead to extracellular matrix remodeling, collagen reorganization, and the rapid formation of a vascular network. A second technology uses ultrasound to control the microstructure of collagen within engineered tissues via a thermal mechanism. Through control of ultrasound heating, engineered tissues can be fabricated with spatial variations in collagen microstructures. The structure of extracellular matrix proteins directs cell behaviors important for tissue formation. In the third technology, ultrasound is used to alter the conformation of another extracellular matrix protein, fibronectin, likely through a cavitation mechanism. These studies highlight how fundamental principles of ultrasound-tissue interactions can be used to develop novel tools for tissue engineering.

2:25–3:10 Break

3:10

1pPA6. Nucleating and sustaining acoustic cavitation for biomedical applications. Tyrone M. Porter (Mechanical Engineering, Boston University, 110 Cummington St, Boston, MA 02215, tmp@bu.edu)

Wes Nyborg was a pioneer in the field of biomedical ultrasound. In particular, Nyborg conducted extensive studies that provided insight into how bubbles oscillating in liquids generated forces that could alter the anatomy and/or physiology of cells. While these studies demonstrated that acoustic cavitation is instrumental in a variety of bioeffects, nucleating and sustaining acoustic cavitation in a controlled manner has proven to be a challenge, particularly in vivo. Studies have shown that these challenges can be addressed with the use of novel materials, particles, and/or acoustic pulsing schemes. In particular, pH-sensitive polymers with varying hydrophobicity have been used to sustain cavitation/cell interactions and vaporizable perfluorocarbon nanoemulsions have been used to reduce the pressure threshold for cavitation nucleation. The impact of these materials and particles on cavitation-mediated bioeffects will be discussed.

3:25

1pPA7. Tracking the motion of cavitation bubbles using pulsed Doppler. E. C. Everbach (Engineering, Swarthmore College, 500 College Avenue, Swarthmore, PA 19081, ceverba1@swarthmore.edu)

Echo-contrast agent microbubbles in blood can sometimes penetrate a clot that is blocking bulk flow in the vessel. When ultrasound is applied for the purpose of sonothrombolysis, microbubbles can be forced by acoustic radiation force into the clot matrix. To monitor the extent of this penetration, a 20 MHz pulsed Doppler method was employed to measure both the position of the bubble front in the clot and its velocity. Correlations between clot dissolution and the location of the advancing microbubble front may be used to optimize cavitational activity and improve sonothrombolysis.

3:40

1pPA8. Recent advances concerning acoustic radiation forces and torques and Wes Nyborg's helpful discussion of acoustic streaming. Philip L. Marston, L. K. Zhang, and David B. Thiessen (Physics and Astronomy Dept., Washington State University, Pullman, WA 99164-2814, marston@wsu.edu)

Recent theoretical advances concerning the geometrical interpretation of acoustic radiation forces [L. K. Zhang and P. L. Marston, Phys. Rev. E 84, 035601R (2011); L. K. Zhang and P. L. Marston, J. Acoust. Soc. Am. 131, EL329-EL335 (2012)] and the scaling of acoustic radiation torques for symmetric objects in beams and in standing waves with increasing helicity [L. K. Zhang and P. L. Marston, Phys. Rev. E 84, 065601R (2011)] will be summarized. For spheres in beams it has been possible to find situations giving transversely stable radiation forces using finite elements. The predicted scaling properties of acoustic torques have been verified in an investigation by an independent group [C. E. M. Demore et al., Phys. Rev. Lett. 108, 194301 (2012); A. G. Smart, Physics Today 65 (6), 18-20 (2012)]. This work will be examined in the context of broader discussions with, and/or a few of the interests of, Wes Nyborg and his practical analysis of acoustic streaming [W. L. Nyborg, in Nonlinear Acoustics, edited by M. F. Hamilton and D. T. Blackstock (Academic Press, San Diego, CA, 1998)] pp. 207-231. [Marston and Thiessen were supported in part by ONR. Zhang was supported in part by NASA.]

3:55

1pPA9. Acoustic streaming in therapeutic ultrasound. Lawrence A. Crum (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, lac@apl.washington.edu)

Wesley Nyborg's contributions to a theoretical description of acoustic streaming were pioneering, rigorous, and so thorough that little additional work has been published. Acoustic streaming has had many applications in acoustics, especially medical acoustics, in which it is difficult to avoid. In many cases, it can be used to enhance or accelerate a particular therapeutic effect. This presentation will provide a few examples of the role of acoustic streaming in therapeutic ultrasound as well as share a few warm memories of this kind and generous man. [Work supported in part by the NIH and NSBRI.]

4:10–4:40 Panel Discussion

Session 1pUW

Underwater Acoustics: Reverberation and Scattering

John R. Preston, Chair
ARL, Pennsylvania State Univ., State College, PA 16804

Contributed Papers

2:30

1pUW1. Planning for a reverberation field experiment. Dajun Tang, Brian T. Hefner, Kevin L. Williams, Jie Yang, and Eric I. Thorsos (Applied Physics Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

A basic research reverberation experiment, supported by the US Office of Naval Research, is planned for 2013. Measurement issues that arise when planning such an experiment are discussed. The fundamental requirement for this basic research experiment is that the environment is characterized in sufficient detail to allow accurate numerical modeling of the acoustical results based on the environmental description. The main goal is to measure mid-frequency shallow water reverberation with full companion environmental measurements so that model/data can be compared without ambiguity. Included in the goal is to make statistical estimates of the uncertainties associated with all the environmental conditions. The frequency range of interest is 1-10 kHz with emphasis at 3 kHz. A pilot field experiment was conducted off the coast of Panama City, Florida. Data from the pilot experiment will be discussed in light of the forthcoming main experiment, including simulations on both propagation/forward-scatter and reverberation for given noise background on both vertical and horizontal arrays which will be deployed in the main experiment.

2:45

1pUW2. Scattering enhancements for partially exposed cylinders at a free surface caused by Franz waves: Measurements and geometric model. Grant C. Eastland and Philip L. Marston (Physics and Astronomy Dept., Washington State University, Pullman, WA 99164-2814, marston@wsu.edu)

Creeping waves on solid cylinders having slightly subsonic phase velocities and large radiation damping are usually described as Franz waves because of their association with complex poles investigated by Franz. For free-field high frequency broadside backscattering in water, the associated echoes are weak due to the radiation damping. It is demonstrated here, however, that for partially exposed solid metal cylinders at a free surface viewed at grazing incidence, the Franz wave echo can be large relative to the specular echo even for ka above 20 when the grazing angle is sufficiently small. The reason is that at small grazing angles and small exposures, the Fresnel zone associated with the specular echo is occluded so that the specular echo is weak [K. Baik and P. L. Marston, IEEE J. Ocean. Eng. 33, 386 (2008)] while the Franz wave is partially reflected at the interface. This has been confirmed from the evolution of echo timing with cylinder exposure and by SAS imaging. In the experiment a solid cylinder was slowly lowered through the free surface into the water. In bistatic scattering a Franz echo can be present with small exposure without the Franz wave reflecting from the free surface. [Supported by ONR.]

3:00

1pUW3. Computation of backscattering enhancements by a half-exposed rigid cylinder at a free surface caused by Franz waves. Anthony R. Smith, Daniel S. Plotnick, Grant C. Eastland, and Philip L. Marston (Physics and Astronomy Dept., Washington State University, Pullman, WA 99164-2814, marston@wsu.edu)

Recent observations of the backscattering by partially-exposed solid aluminum cylinders in water viewed at grazing incidence at a free surface [G. C. Eastland, Ph.D. thesis, Wash. State Univ., 2012] indicate that the generation, propagation, and reflection of Franz-type creeping waves can be important. The present investigation gives additional support for this hypothesis by calculating the exact backscattering by a half-exposed infinitely long rigid cylinder viewed over a range of grazing angles. The calculation begins with the known frequency domain expression for the complex amplitude given in an Appendix of [K. Baik and P. L. Marston, IEEE J. Ocean. Eng. 33, 386 (2008)]. Numerical Fourier transforms were used to construct the time-domain response for various excitations and the evolution of that response was investigated as a function of the grazing angle. This procedure reveals from the timing of the computed features there is a significant delayed echo having the expected timing of a Franz wave partially reflected from the free surface. The timing of the Franz wave depends on grazing angle in agreement with a geometric model in Eastland's thesis. [Supported by ONR.]

3:15

1pUW4. Some initial findings from the very shallow water GulfEx12 reverberation experiments. John R. Preston (ARL, Penn State Univ., P.O. Box 30, MS3510, State College, PA 16804, jrp7@arl.psu.edu)

In April 2012, reverberation and clutter measurements were taken in very shallow water (~20 m) over a 40 hour period off Panama City, FL. This work describes the data from this recent sea trial called GULFEX12 designed to characterize reverberation and clutter from a very shallow water site in the 2500-5500 Hz band. The received data are taken from one aperture of the Five Octave Research Array (FORA) namely, the triplet sub-aperture. The array was fixed 2 m off the sea floor by mounting it to tripods using a clothes line and data were passed by cable to a nearby moored ship (the R/V Sharp). An ITC source transducer was located 3 m away from the array center. Data show a surprising amount of anisotropy. Five different pulses were used in this study. Matched filtered polar plots of the reverberation and clutter are presented using the FORA triplet beamformer to map out anisotropy. Some model data comparisons are made using the author?s normal mode based reverberation model. Help from D.J. Tang, T. Hefner and K. Williams of Applied Physics Lab at Univ. of Washington was crucial to this effort. [Work supported by ONR code 322OA.]

3:30

1pUW5. Scattering of low-frequency spherical waves by fluid and solid spheres. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Research Laboratory, Physical Sciences Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

Acoustic Green's functions for a homogeneous fluid with an embedded spherical obstacle arise in analyses of sound scattering by air bubbles, scattering by objects on or near the seafloor, radiation by finite sources, sound attenuation in and scattering from clouds of suspended particles, etc. Here, radius of the obstacle is assumed to be small compared to the wavelength of sound. This regime is usually referred to as Rayleigh scattering. A new, elementary solution of the problem of diffraction of a spherical wave was recently obtained for small, soft obstacles [O. A. Godin, J. Acoust. Soc. Am. **37**, L13605 (2010)]. The solution is valid for arbitrary positions of the source and receiver relative to the obstacle. In this paper, the solution is extended to homogeneous and inhomogeneous fluid and solid spheres. Low-frequency scattering is found to be rather sensitive to boundary conditions on the surface of the obstacle. Resonance scattering of spherical waves by small spheres is investigated. [Work supported, in part, by ONR.]

3:45

1pUW6. Fast model for target scattering in a homogeneous waveguide. Steven G. Kargl, Kevin L. Williams, and Aubrey L. Espana (Applied Physics Laboratory, University of Washington, 1013 NE 40th St, Seattle, WA 98105, kargl@apl.washington.edu)

A fast ray model for propagation in a homogenous water column tracks time-of-flight wavepackets from sources to targets and then to receivers. The model uses image sources and receivers to account for interactions with the water column boundaries, where the layer of water lies between an upper semi-infinite halfspace of air and a lower semi-infinite halfspace of a homogenous sediment. The sediment can be either an attenuating fluid with a frequency-independent loss parameter or a fluid consistent with an effective density fluid model (i.e., a fluid limit to Biot's model for a fluid-saturated poroelastic medium). The target scattering process is computed via convolution of a free-field scattering form function with the spectrum of an incident acoustic field at the target location. A simulated or measured scattered free-field pressure from a complicate target can be reduced to a scattering form function, and this form function then can be used within model via interpolation. The fast ray-based model permits the generation of sets of realistic pings suitable for synthetic aperture sonar processing for proud and partially buried target. Results from simulations are compared to measurements where the targets are an inert unexploded ordnance and aluminum cylinder. [Research supported by SERDP and ONR.]

4:00

1pUW7. Supervised machine learning for classification of underwater target geometry from sampled scattered acoustic fields in the presence of rough bottom interference. Erin M. Fischell and Henrik Schmidt (Mechanical Engineering, MIT, 77 Massachusetts Ave., Cambridge, MA 02139, emf43@mit.edu)

An increasingly important mission for Autonomous Underwater Vehicles (AUVs) is the identification and classification of potentially hazardous targets in harbors. A process is demonstrated in simulation that would allow AUV in-flight classification of spherical and cylindrical targets using only scattered acoustic amplitudes collected at waypoints. Target and bottom roughness scattered fields are simulated using OASES and SCATT, then combined and sampled into independent training and testing examples for a Support Vector Machine (SVM). The feature space and parameters for the SVM are selected using a design of experiments grid search. By processing the model using a feature reduction algorithm, it is possible to identify the regions in the scattered field that are the most critical for classification. To make use of the resulting models and critical features, a vehicle in the field would be loaded with pre-generated models for bottom and target classification. Upon target localization, the vehicle would begin visiting the critical waypoints until a confident classification is achieved using the SVM models. The resulting in-flight classification, based only on amplitude data collected by a hydrophone along the vehicle's path, could be used as the basis for further action on the target. [Work supported by ONR and NSF GRFP.]

4:15

1pUW8. Simulating coherent backscatter enhancement from aggregations of underwater scatterers. Adaleena Mookerjee and David R. Dowling (Mechanical Engineering, University of Michigan, Ann Arbor, MI 48105, adaleena@umich.edu)

Remote classification of scatterers is an enduring priority in active sonar applications. For aggregations of marine life, typically schools or shoals of fish, remote classification may be possible when there is coherent backscattering enhancement (CBE) from the aggregation. CBE is a multiple scattering phenomenon that occurs in optics and acoustics when wave propagation paths within the aggregation are likely to be traversed in both directions. For a plane wave illuminating a half-space of randomly-placed, omnidirectional scatterers, CBE may lead to a doubling of the scattered field intensity in the direction opposite that of the incident wave (Akkermans et al. 1986). This presentation describes acoustic CBE simulations for finite-size spherical aggregations of point scatterers. The results are developed from the classical multiple-scattering equations of Foldy (1945), and are checked to ensure appropriate rotational symmetries and acoustic energy conservation when scatterer locations are random and when they are structured. Variations in the magnitude of the CBE peak and its angular width are presented for different frequencies, aggregation sizes, scatterer densities, and scatterer properties. Extension of these results to sonar-pulse scattering from schools of fish will be discussed. [Supported by the Office of Naval Research.]

4:30

1pUW9. Frequency response of ordnance replicas across multiple scales. Christopher Dudley, Jermaine Kennedy, Kwang Lee, and David Malphurs (Naval Surface Warfare Center, Panama City Division, 110 Vernon Ave, Panama City, FL 32407, mhhd@hotmail.com)

Broad-band, multi-aspect backscatter data obtained using small-scaled and full-scaled replicas of Unexploded Ordnance (UXO) are reported. Data were collected using either linear or circular rail systems. The experiments were performed in: (1) NSWC PCD's small scale test bed (less than 1/16 scaled) which has a simulated bottom, (2) NSWC PCD's Facility 383 9-million gallon fresh water test pool which has a 5-ft sand bottom, and (3) the Gulf of Mexico during GULFEX12 off Panama City, Florida. Data were processed using linear and circular synthetic aperture sonar techniques to generate both images and plots of target strength as functions of frequency and aspect angle. The results from all three experiments are compared to each other and with predictions from Finite-element (FE) analysis. These comparisons are used to assess the utility of alternative methods for generating sonar data from bottom targets of sufficient fidelity to study scattering phenomena and support development of automated target recognition. [Work supported by ONR Code 32 and SERDP.]

Payment of additional registration fee required to attend. See page A?

MONDAY EVENING, 22 OCTOBER 2012 COLONIAL, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on the Acoustics of Pianos

James P. Cottingham, Chair
Physics, Coe College, Cedar Rapids, IA 52402

Chair's Introduction—7:00

Invited Paper

7:05

1eID1. The acoustics of pianos. Antoine J. Chaigne (ENSTA, UME, Chemin de la Huniere, Palaiseau, 91761, France, antoine.chaigne@ensta-paristech.fr)

The manufacturing of pianos remains largely empirical, with numerous trial-and-error procedures and fine adjustments at each step of the building process. The "skeleton" of the instrument obeys fundamental principles of vibrations, acoustics, and material science. An abundance of literature is available on its different constitutive parts. However, scientific studies based on a global model of the instrument that connects all of these constitutive parts together are more recent. Such modeling sheds useful light on the essential coupling properties between elements and, in particular, on the string-soundboard coupling at the bridge, and on the radiation of the soundboard. Fine analysis of piano tones also shows that in most cases, a nonlinear model of the strings is necessary to account for perceptually significant features such as precursors in the time-domain and the so-called "phantom partials" in the spectrum. This nonlinearity is based on the coupling between transverse and longitudinal waves in the string. In this lecture, a time-domain model of a complete piano is presented that couples together nonlinear strings, soundboard vibrations, and radiation in air. It highlights the transmission of both transverse and longitudinal string forces to the soundboard, and the influence of rib design and bridge on soundboard mobility and radiation patterns. Comparisons between the results of the model and measurements made on real pianos will be discussed.

Session 2aAA

Architectural Acoustics, Psychological and Physiological Acoustics, Engineering Acoustics, and Noise: Binaural and Spatial Evaluation of Acoustics in Performance Venues

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Troy, NY 12180

David H. Griesinger, Cochair
Research, David Griesinger Acoustics, Cambridge, MA 02138

Chair's Introduction—7:55

Invited Papers

8:00

2aAA1. Modeling binaural processing: What next? Jens Blauert (Ruhr-University Bochum, Communication Acoustics, Bochum 44780, Germany, jens.blauert@rub.de)

Models of binaural processing are traditionally based on signal-processing in a bottom-up (signal-driven) architecture. Such models are sufficient for a number of technologically important applications, such as perceptual coding, sound-source identification and localization, dereverberation and decoloration, but fail when applications require cognition. Future models will thus include symbol processing in addition to signals processing, will have inherent knowledge bases and employ top-down (hypothesis-driven) strategies in addition to bottom-up ones. Some of these features are known from automatic speech recognition and may be generalized for broader application, e.g., blackboard structures. With the new models more sophisticated applications may be approached, for instance, quality evaluation and assessment on the basis of internal references, such as needed to determine estimates of the quality of performance spaces and/or audio systems. Further, to enable autonomous learning, future models will employ feed-back loops to realize active exploratory actions. Some of these features can be imported from recent research in robot audition. In our contribution, we shall, among other things, report on ideas and concepts as currently discussed in AABBA, an international grouping of 14 laboratories in Europe and the US that are dealing with auditory assessment by means of binaural algorithms.

8:20

2aAA2. Sound quality from binaural and multidirectional measurements. David H. Griesinger (Research, David Griesinger Acoustics, 221 Mt Auburn St #504, Cambridge, MA 02138, dgriesinger@verizon.net)

There has been rapid progress in methods to gather binaural and multi-directional point-to-point impulse responses from unoccupied venues. With individual matching of headphones to a listener this data can sometimes be auralized to obtain a glimpse of the sound of a venue. But numerical methods to analyze such data in order to quantify the precise sound of a particular seat remain elusive. This paper will discuss some of the limitations of point measurements, recent work on binaural technology, and impulse response analysis techniques that are not based on sound energy, but on the methods used by the ear and brain to aurally perceive music, speech, and sonic environments. The goal - nearly in sight - is developing methods for obtaining objective quality assessments from binaural or multidirectional recordings of live music and speech.

8:40

2aAA3. Modeling binaural suppression processes for predicting speech intelligibility in enclosed spaces. Vanessa Li (Graduate Program in Architectural Acoustics, School of Architecture, Rensselaer Polytechnic Institute, Troy, NY 12180, vanessa.li@gmail.com), Ning Xiang, and Jonas Braasch (Graduate Program in Architectural Acoustics, School of Architecture, Rensselaer Polytechnic Institute, Troy, NY)

Speech from a target speaker reaches a listener via multiple paths in a room due to room reflections. As sound waves from the target speaker approach the listener, degradation to the signal is caused by ambient noise and reverberant energy. The speech transmission index (STI) is a commonly used metric for predicting speech intelligibility accounting for both noise and reverberation. This metric, however, is a monophonic measure that does not take into consideration binaural cues used for unmasking undesired effects. As a result, using the STI on its own tends to under-predict intelligibility under binaural listening conditions. The proposed research aims to improve speech intelligibility predictions with the presence of room effects by implementing a psychophysical binaural model as a front-end to the STI calculation. The equalization-cancellation (EC) theory is applied to spatially unmask noise, while late incoherent reverberant energy is suppressed by applying a weighting function based on interaural coherence. Preliminary comparisons between listening tests and model predictions reveal promising results, indicating a useful tool in acoustical planning in addition to further study into binaural suppression processes.

9:00

2aAA4. Utilizing head movements in the binaural assessment of room acoustics and analysis of complex sound source scenarios. Jonas Braasch, Anthony Parks, Torben Pastore, and Samuel W. Clapp (School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180, braasj@rpi.edu)

The measurement of transfer functions is currently standard practice for the acoustic evaluation of performance venues. The pathway between a measurement loudspeaker and a microphone or binaural manikin in a room can be treated as a linear time-invariant system, and meaningful acoustical parameters can be derived from measured impulse responses. Unfortunately, this method neglects that human listeners typically move their heads when exploring an acoustic venue. This paper addresses these implications when designing systems to take head movement into account. A number of approaches will be discussed based on existing research and technology at Rensselaer, including a binaural manikin with a motorized head, a technique to simulate head movements from impulse responses recorded with a higher-order spherical microphone, and a binaural model that can process head movements. The model distinguishes between a room coordinate system and a head-related coordinate system. Its binaural activity map is rotated with head movements in order to separate front-back images, resolve the reverberation-reduced angle of lateral sound sources, assess different surround loudspeaker configurations for immersive sound systems, and separate acoustic sources. The research presented here has received support from the National Science Foundation under Grant No. 1002851.

9:20

2aAA5. Using a higher-order spherical microphone array to assess spatial and temporal distribution of sound in rooms. Gary W. Elko (mh Acoustics LLC, 25A Summit Ave, Summit, NJ 07901, gwe@mhacoustics.com) and Jens M. Meyer (mh Acoustics LLC, Fairfax, NJ)

We have developed a spherical microphone called the Eigenmike® microphone array that is capable of achieving up to third-order spherical harmonics decomposition of the sound field. One potential use of the spherical array is to investigate the spatial nature of sound fields in rooms. In this talk, we will show some measurement results where the processed data from an Eigenmike® array is used to compute various energy ratios between the direct, lateral, rear, and floor and ceiling directions. We will also show some other simple measures that might be useful in the spatial analysis and characterization of room acoustics.

9:40

2aAA6. Reciprocal binaural room impulse response measurements. Johannes Klein, Martin Pollow, Janina Fels, and Michael Vorlaender (ITA, RWTH Aachen University, Neustr. 50, Aachen 52066, Germany, mvo@akustik.rwth-aachen.de)

Multi-channel spherical loudspeakers have been introduced in shapes of cubes, dodecahedra, or higher-order discrete representations of spheres. In this contribution a spherical source with a partial Gaussian distribution of 28 channels is presented. With sequential measurements and rotation of the sphere a radiation of effectively 23rd order of spherical harmonics can be obtained. Accordingly directional patterns of not only sound sources but also of receivers such as HRTF can be modeled in detail up to quite high frequencies. The high order of spherical harmonics allows investigation of individual differences of pinna cues. When applied in a reciprocal measurement of room impulse responses in performance venues, an almost perfect omnidirectional microphone on the stage and an HRTF source in the audience can be used to study spatial room acoustic parameters such as early lateral energy fractions, late lateral strength and IACC of dummy heads and individuals. This is obtained by post processing of just one set of multi-channel impulse responses in the venue. Opportunities and challenges of this approach will be discussed.

10:00–10:20 Break

10:20

2aAA7. Measuring and inferring the directional properties of the early room response. Jonathan Botts, Samuel Clapp, Ning Xiang, and Jonas Braasch (Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th St., Greene Building, Troy, NY 12180, botts.jonathan@gmail.com)

In an effort to understand the response of a room more completely, spherical microphone arrays have been used to produce a three-dimensional map of a room impulse response. To locate reflections from various room surfaces, the most common approach is to search for peaks in the beamformed response. Particularly when using this approach with a low-order array there is no way to distinguish a side-lobe from an additional arrival. Furthermore, overlapping arrivals skew the maxima of beampatterns, resulting in incorrect inferences. This talk seeks to demonstrate that a Bayesian, model-based analysis of the data addresses the complete problem of image source estimation, with mechanisms to determine the number and locations of simultaneous arrivals. Particularly with low-order arrays, substantially more accurate estimates can be made, which both increases the overall quality of the analysis and extends the portion of the impulse response that may be reliably analyzed.

10:40

2aAA8. A measurement technique achieving high spatial resolution for sound sources within a performance venue. Alex Case (Sound Recording Technology, University of Massachusetts, Lowell, MA 01854, alex@fermata.biz), Agnieszka Roginska, and Jim Anderson (New York University, New York, NY)

A proof of concept for gathering high spatial resolution sound radiation, from near field to far field, in 3 dimensions around an electric guitar amplifier is presented, with an eye and ear toward applying a similar technique to other essential sound sources. A high density microphone array is used to gather many thousands of impulse response in a hemi-anechoic space. The resulting data serves as a useful input to room models and auralizers, but finds added purpose as an educational tool in musical acoustics and sound recording.

11:00

2aAA9. Comparison of headphone- and loudspeaker-based concert hall auralizations. Samuel Clapp (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Greene Building, Troy, NY 12180, clapps@rpi.edu), Anne Guthrie (Arup Acoustics, New York, NY), Jonas Braasch, Ning Xiang, and Terence Caulkins (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Troy, NY)

In this research, a spherical microphone array and a dummy head were used to measure room impulse responses in a wide variety of concert and recital halls throughout New York State. Auralizations were created for both headphone playback and second-order ambisonic playback via a loudspeaker array. These two systems were first evaluated objectively to determine the level of accuracy with which they could reproduce the measured soundfields, particularly with respect to important binaural cues. Subjects were then recruited for listening tests conducted with both reproduction methods and asked to evaluate the different spaces based on specific parameters and overall subjective preference, and the results of the two playback methods were compared.

11:15

2aAA10. A spatial encoding method for measured room impulse responses. Sakari Tervo, Jukka Pätynen, and Tapio Lokki (Department of Media Technology, Aalto University School of Science, P.O. Box 15500, Espoo FI00076 Aalto, Finland, sakari.tervo@aalto.fi)

The spatial information contained in measured room impulse responses can be used to explain some of the acoustical properties of performance spaces. This paper presents a spatial encoding method, which can extract accurate spatial information from impulse responses that are measured with at least four microphones in an open 3-D array. The method is based on decomposing a spatial room impulse response into a set of image-sources, i.e., every single sample in the impulse response is considered as an image source. Each of the image-sources is localized with an acoustic source localization method, which depends on the applied microphone array and the acoustic conditions. Due to the image-source presentation, the presented method can be applied to any compact array and used in conjunction with variety of current spatial loudspeaker reproduction systems to create convolution reverb-type spatial sound reproduction. The method allows static and interactive binaural reproduction via virtual loudspeaker arrays. The presentation includes demonstrations with a binaural reproduction system.

11:30

2aAA11. Source locations, listener locations, and measurement devices when making acoustical measurements in performance spaces. Elizabeth Lamour (University of Kansas, Lawrence, KS 66045, lizlamour@gmail.com)

Does the location of the source affect the results of an acoustical measurement? This is the question that sparked the author's Master's Project which explores the differences between measurements taken with a source located on the stage of a performance space and a source located higher above the stage using the space's existing sound system. Impulse responses were gathered from four different performance halls with respect to source location, microphone location, and measurement devices used. Comparisons were made between trends in reverberation time, early decay time, and interaural cross-correlation coefficient. The results are not only interesting, but they also question the typical measurement practices of acousticians and confirm assumptions made regarding important acoustical characteristics of performance spaces.

11:45

2aAA12. Applying direct algebraic sound source localization method for time-domain reflectometry of conference room. Tsukassa Levy and Shigeru Ando (Information Physics and Computing, Tokyo University, Tokyo-to Bunkyou-ku Hongou, Tokyo 113-0033, Japan, levy.t@alab.t.u-tokyo.ac.jp)

A novel localization method has been previously exposed, using an explicit formula of direction and distance of a monopole source and a circular array [S. Ando, ASA Seattle Meeting, 2011]. However, this localization method has little been applied in real environments, such as concert halls or conference rooms. It has also been shown that the algorithm has a high temporal resolution [T.Levy, S. Ando, Hong Kong Acoustics 2012, 2012] that enables to localize sound sources using reflected sound waves. Thus, the aim of this study is to study conference room's reflection using the proposed algorithm and to test its robustness and its efficiency in such conditions. In the experiments, the main reflectors in the conference room will be identified and a comparison between the previously proposed method and the traditional sound source localization algorithms is done, in terms of rapidness and precision.

Session 2aAB

Animal Bioacoustics and Acoustical Oceanography: Acoustics as Part of Ocean Observing Systems

Ana Sirovic, Cochair
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Lora J. Van Uffelen, Cochair
Ocean and Resources Engineering, University of Hawaii at Manoa, Honolulu, HI 96815

Chair's Introduction—7:55

Invited Papers

8:00

2aAB1. Acoustic tomography as a component of the Fram Strait observing system. Brian D. Dushaw, Hanne Sagen, Stein Sandven (Nansen Environ. and Remote Sensing Ctr., N-5006, Norway, brian.dushaw@nersc.no), and Peter Worcester (Scripps Institution of Oceanography, UCSD, La Jolla, CA)

The Fram Strait, a deep constriction between Svalbard and Greenland, is the primary location for the exchange of heat, mass and freshwater between the Arctic and Atlantic Oceans. With existing data and ocean modeling, current estimates of these exchanges, critical for understanding the Arctic Ocean climate, are inaccurate. To try to improve these estimates, during 2008-9 the DAMOCLES project deployed a test tomography path spanning the deep, ice-free part of the northward-flowing West Spitzbergen Current (WSC). Small-scale scintillations of sound speed due to eddies, fronts, and internal waves, are an important aspect of acoustic propagation of the region. Variability within Fram Strait, and the WSC in particular, is characterized by ubiquitous mesoscale eddies with 20-km scale. These eddies extend to depths of several hundred meters. Understanding the forward problem is essential for the inversion of acoustic data. The sound speed environment of Fram Strait generally prevents individual ray arrivals from being resolved in O(100-km) acoustic paths. An accurate inversion of these data for path-averaged sound speed (temperature) can be still be obtained, however. An objective mapping study, combining acoustic and existing data types, demonstrates that tomography will be a valuable and effective addition to the Fram Strait observing system.

8:20

2aAB2. Long-term acoustic monitoring off Central California. Curtis A. Collins, John E. Joseph, Ching Sang Chiu, John Colosi, and Christopher W. Miller (Oceanography, Naval Postgraduate School, 833 Dyer Rd, Monterey, CA 93943-5122, collins@nps.edu)

The use of moored ocean arrays for environmental acoustic measurements off Central California is discussed. A cabled array off Point Sur, CA, which was designed for long-range, low-frequency listening was used by NPS and collaborators from late 1997 through mid-2000 and provides examples of a wide range of activities including use for student laboratories, faculty and student research, as well as monitoring, e.g. ambient acoustic noise, test ban treaty activities. From mid-2006 to present, passive acoustic data have continued to be collected off Pt. Sur using single hydrophone moored autonomous listening stations which record data intermittently at sampling rates of 200 kHz. We have recently considered re-establishment of cabled passive acoustic measurements using MARS, an example of an observatory which was designed and located for more traditional oceanographic studies. The utility of MARS for acoustic measurements depends both on how well it can characterize the regional acoustic environment as well as local oceanographic processes that can be resolved acoustically (canyon effects, geography, sound speed variability, sediments and local vessel traffic). These can be contrasted with existing cabled and autonomous data from Point Sur.

8:40

2aAB3. Acoustics at the ALOHA Cabled Observatory. Bruce M. Howe (Ocean and Resources Engineering, University of Hawaii at Manoa, 2540 Dole St, Holmes Hall 402, Honolulu, HI, bhowe@hawaii.edu), Fred Duennebier (Geology and Geophysics, University of Hawaii at Manoa, Honolulu, HI), Roger Lukas (Oceanography, University of Hawaii at Manoa, Honolulu, HI), and Ethan Roth (Ocean and Resources Engineering, University of Hawaii at Manoa, Honolulu, HI)

Since 6 June 2011, the ALOHA Cabled Observatory (ACO) has been collecting ocean acoustic data, continuing an earlier data set covering February 2007 - October 2008. The ACO is at Station ALOHA 100 km north of Oahu, the field site of the Hawaii Ocean Time-series (HOT) program that has collected biological, physical, and chemical oceanographic data since 1988. At 4728 m water depth, it is the world's deepest operating cabled observatory. ACO provides power and communications to user instrumentation. Among the instrumentation there are two hydrophones 1 m off the bottom separated by 1 m. One is an OAS Model E-2PD meant for low frequencies (0.014 Hz to 8 kHz). A second (uncalibrated) hydrophone is meant for higher frequencies. Current sampling rates for both hydrophones are 96 kHz; subsampled 24 kHz data are streamed to the Web in real-time. The system will be described and examples of acoustic events and signals presented, including local and distant earthquakes, marine mammals, surface waves, wind, rain, ships, sonars, and implosions. Plans for future acoustics research will be discussed. [Work supported by the National Science Foundation.]

9:00

2aAB4. On the region of feasibility of interference alignment for underwater acoustic communication. Dario Pompili (Electrical and Computer Engineering, Rutgers University, 64 Brett Road, Piscataway, NJ 08854, pompili@cac.rutgers.edu)

To enable underwater applications such as coastal and tactical surveillance, undersea explorations, and picture/video acquisition, there is a need to achieve high data-rate underwater acoustic communications, which translates into attaining high acoustic channel spectral efficiencies. Interference Alignment (IA) technique, which has recently been proposed for radio-frequency MIMO terrestrial communication systems, aims at improving the spectral efficiency by enabling nodes to transmit data simultaneously at a rate equal to half of the interference-free channel capacity. The core of IA lies in designing transmit precoding matrices for each transmitter such that all the interfering signals align at the receiver along a direction different from the desired signal. To decode, the receiver projects the received signal onto a decoding vector that is orthogonal to the data vector of interfering signal. While promising, there are challenges to be solved for the use of IA underwater, i.e., 1) imperfect acoustic channel knowledge, 2) high computational complexity, and 3) high communication delay. We study the feasibility of IA in underwater environment under these challenges; we also propose a distributed computational framework to parallelize the iterative IA algorithm and determine to what extent we can parallelize it among neighboring nodes under different channel coherence times.

9:20

2aAB5. Recent development and application of active acoustic techniques for studies of zooplankton ecology and implications for ocean observatories. Gareth L. Lawson, Andone C. Lavery, Peter H. Wiebe, Jonathan R. Fincke, and Nancy J. Copley (Woods Hole Oceanographic Institution, 266 Woods Hole Rd, Woods Hole, MA 02543, glawson@whoi.edu)

High-frequency active acoustic techniques enjoy a long history in the study of zooplankton ecology and increasingly are being incorporated into ocean observing systems, addressing a pressing need for zooplankton-sampling capabilities. Discriminating among sources of scattering remains a key problem in ecological applications of active acoustics, however, especially when deploying on autonomous platforms, where independent sampling with nets or optics to verify acoustic observations is often not feasible. Here we consider the ecological insights that can be afforded by active acoustic methods and implications to the design of ocean observing systems by reporting on (1) a series of recent field studies of krill ecology employing both a traditional multi-frequency system (43, 120, 200, 420 kHz) and a recently-developed broadband system (30-600 kHz) designed to provide enhanced capabilities for discrimination of scattering sources, and (2) test deployments on autonomous platforms of a low-power active acoustic system capable of broadband or narrowband transmission. Comparisons to concurrent sampling with a depth-stratified net system, when available, allow an assessment of the abilities of these acoustic systems for remotely discriminating among sources of scattering and for estimating the abundance and size of animals.

9:40

2aAB6. Passive acoustics monitoring as part of integrated ocean observing systems. Joseph J. Luczkovich (Biology/Institute for Coastal Science and Policy, East Carolina University, 383 Flanagan Building, Greenville, NC 27858, luczkovichj@ecu.edu), Mark W. Sprague (Physics, East Carolina University, Greenville, NC), Cecilia S. Krahforst (Coastal Resources Doctoral Program, East Carolina University, Greenville, NC), D. Reide Corbett, and John P. Walsh (Geological Sciences & Institute for Coastal Science and Policy, East Carolina University, Greenville, NC)

Passive acoustic monitoring can be a useful tool to include on Ocean Observing Systems. As an example, we describe the monitoring the acoustic environment in the coastal waters of North Carolina (USA) using an instrumented platform. The ECU Itpod (instrumented tripod) has been deployed in several locations in Pamlico Sound and river estuaries since 2006 to study fishes in the Family Sciaenidae (drums and croakers). We will present data recorded with hydrophones deployed on the Itpod with remote data loggers, acoustic Doppler current profilers, turbidity meters and water quality instruments. We have used passive acoustic recordings to study the correlations of fish sounds and environmental parameters (temperature, salinity, turbidity, dissolved oxygen, wave action, river discharge, tropical storms). The long-term data suggest that spring temperature increases are associated with increased activity of acoustically mediated courtship and spawning behavior of sciaenid fishes; these sounds decline in the fall as water temperature declines. In addition, we have observed acoustic interactions between marine mammal predators and their fish prey and the effects of noise from tugs and small boats on fish sound production. Itpods must be recovered periodically to recover data and replenish batteries; solar-powered platforms and automated fish detection algorithms are under development.

10:00–10:30 Break

10:30

2aAB7. The power of acoustics in ocean observing systems: A case study in the Bering Sea. Jennifer L. Miksis-Olds and Laura E. Madden (Applied Research Laboratory, The Pennsylvania State University, PO Box 30, State College, PA 16804, jlm91@psu.edu)

Acoustic time series are incredibly powerful as independent data sets. Passive acoustic recordings provide information on environmental sound levels, the presence of vocalizing animals, surface conditions, marine precipitation, and anthropogenic activities within the area of acoustic coverage. Active acoustic systems provide a time series of acoustic backscatter from which biological scatter can be measured and quantified to provide estimates of relative abundance and numerical density. The combination of acoustic technology with other hydrographic sensors within an ocean observing system now affords the opportunity to develop an understanding of ecosystem dynamics ranging from the physical oceanographic conditions to the distribution and behavior patterns of top predators. This is especially critical in sub-Arctic regions like the Bering Sea where rapid changes associated with climate change are having impacts at multiple levels. Here we discuss the environmental parameters that are the best predictors of different marine mammal species as determined through generalized linear and general additive mixed models. Predictor variables considered were percent ice cover, ice thickness, sound level at five frequencies, and percent composition of 4 biologic scattering groups.

10:50

2aAB8. A low cost, open source autonomous passive acoustic recording unit for recording marine animals. Robert D. Valtierra (Dept. Mechanical Engineering, Boston University, 110 Cummington St., Boston, MA 02215, rvaltier@bu.edu), Sofie M. VanParijs (Northeast Fisheries Science Center, National Oceanic and Atmospheric Administration, Woods Hole, MA), R. G. Holt, Connor Mace, Kara Silver, and Chris Bernard (Dept. Mechanical Engineering, Boston University, Boston, MA)

An autonomous passive acoustic recording unit (ARU) was developed through a collaboration between the Boston University Department of Mechanical Engineering and the NOAA Northeast Fisheries Science Center. The ARU consists of two main sections, an electronic data logger and a mechanical pressure case and release. The datalogger makes use of widely adopted commercial hardware such as SD card memory and USB connectivity. In addition, WAV file formats and open-source compiler software allow flexibility and programmability at minimal expense. The pressure case was designed for shallow water (100 m) applications with few machined parts and several "off the shelf" parts. The overall system can be constructed at a minimal cost and has been successfully tested during both laboratory and at-sea trials.

11:05

2aAB9. Autonomous detection of neotropical sciaenid fishes. Sebastian Ruiz-Blais, Mario R. Rivera-Chavarria (Centro de Investigaciones en Tecnologías de la Información y Comunicación, Universidad de Costa Rica, Sede "Rodrigo Facio Brenes" Montes de Oca, San José 2060, Costa Rica, mariorrivera@gmail.com), and Arturo Camacho (Escuela de Ciencias de la Computación en Informática, Universidad de Costa Rica, San Jose, Costa Rica)

Sciaenid passive acoustics are a demonstrated valuable tool for fisheries management. In spite of this, an efficient software tool to detect and identify fish sounds is not currently available. Such tool would be useful for autonomous recognition and array methodologies. For Neotropical environments this lack is even more conspicuous since the availability of corroborated sciaenid sounds is limited. We are developing such tools using corroborated Cynoscion squamipinnis (Pisces: Sciaenidae) sounds. Our approach is based on timbre statistics, short and long-term partial loudness, and the 30 Hz typical pattern found on the signal's stridulations. Relevant fish drums are detected through empirically found fix thresholds for the timbre statistics and the 30 Hz pattern, and a dynamic threshold established by an unsupervised algorithm based on the long-term loudness. Current results show a recognition rate of 80%. Despite these promising numbers, there are still challenges ahead. In the future, we plan to incorporate other variables that affect underwater sound characteristics such as depth, source level distance, and physical chemical properties, which may be crucial to make a user friendly, accurate, and practical tool, for neotropical marine environmental managers. We also plan to extend this method to other soniferous coastal fish.

11:20

2aAB10. Fish recordings from NEPTUNE Canada. Ana Sirovic, Sophie Brandstatter, and John A. Hildebrand (Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, CA 92093-0205, asirovic@ucsd.edu)

NEPTUNE Canada is a regional-scale ocean observing system deployed off the west coast of Vancouver Island, Canada. Among the data streams broadcast live over the internet are video collected using black and white low-light camera and audio collected with Naxys hydrophone (5 - 3,000 Hz). These data allow for description of sound production by fishes in the vicinity of the system. Concurrent video and hydrophone data are available from the Barkley Canyon node (~900 m depth). While the hydrophone recordings were continuous, strobes for video are only turned on during short, irregular (~10 min) intervals. Approximately 30 h of concurrent video and audio recordings were analyzed. The most commonly seen fish was sablefish (*Anoplopoma fimbria*), and the most common fish-like sound was a broadband, short pulse that occurred on nearly half of the recordings. On approximately one-fifth of concurrent video and audio recordings both sablefish and fish-like pulsed sounds were detected. It may be possible to use these sounds to monitor sablefish abundance across the northeastern Pacific Ocean.
NEPTUNE Canada Data Archive, http://www.neptunecanada.ca, hydrophone and video data from May, June, August, and December 2010 and January and February 2011, Oceans Networks Canada, University of Victoria, Canada. Downloaded 2012.

11:35

2aAB11. Tracking and source level estimation of multiple sperm whales in the Gulf of Alaska using a two-element vertical array. Delphine Mathias, Aaron M. Thode (Marine Physical Lab, Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, CA 92037-0238, delphine.mathias@gmail.com), Jan Straley (University of Alaska Southeast, Sitka, CA), and Russel D. Andrews (School of Fisheries and Ocean Sciences, University of Alaska Fairbanks, Fairbanks, AK)

Between 15-17 August 2010 a two-element vertical array (VA) was deployed in 1200 m deep water off the continental slope of Southeast Alaska. The array was attached to a longline fishing buoyline at 300 m depth, close to the sound-speed minimum of the deep-water profile. The line also attracted seven depredating sperm whales to the area, each generating impulsive 'clicks' that arrived on the VA via multiple ray paths. The propagation model BELLHOP was used to model relative arrival times and vertical elevation angles of click ray paths as a function of depth and range from the VA. The resulting tracking system yielded range-depth tracks of multiple animals out to at least 35 km range. These locations, along with the transmission loss estimates of the model, permitted the sound source levels to be recovered. Here we present the consistency of source levels from individuals over time, the degree of source level variation between individuals, and possible correlations between inter-click interval and source level. This analysis suggests how a relatively simple ocean observing acoustic system could localize bioacoustic signals over large ranges, given the appropriate deployment configuration.

11:50

2aAB12. Acoustic thermometry as a component of the global ocean observing system. Brian D. Dushaw (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105-6698, dushaw@apl.washington.edu)

Acoustic data acquired during the 1995-2006 Acoustic Thermometry of Ocean Climate (ATOC) program were used to test the accuracy of ocean state estimates of the North Pacific obtained by various means: simple forward integration of a model, objective analysis of hydrographic and altimeter data, and data assimilation using general circulation models. The comparisons of computed and measured time series stringently tested the accuracy of the state estimates. The differences were substantial, indicating that acoustic thermometry provides unique information about the large-scale temperature. On some acoustic paths, changes in temperature occurring over time scales of weeks with magnitudes comparable to the seasonal cycle were observed. Acoustic thermometry offers valuable constraints on the large-scale thermal variability for the ocean observing system. Acoustic tomography was accepted as part of the Ocean Observing System during the OceanObs'99 and '09 international workshops. Sources and receivers of acoustic thermometry can serve multiple purposes. Hydrophone arrays are used to study a wide range of human, biological, and geological activity. Acoustic sources can transmit signals that can be used to track drifting instrumentation. A modest number of active and passive acoustic instruments deployed worldwide can form a general purpose global acoustic observing network.

Session 2aBA

Biomedical Acoustics and Physical Acoustics: Modeling of Nonlinear Medical Ultrasound

Martin D. Verweij, Cochair
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Koen W.A. van Dongen, Cochair
Acoustical Wavefield Imaging, Delft University of Technology, Delft 2600 GA, Netherlands

Chair's Introduction—8:00

Invited Papers

8:05

2aBA1. Full-wave nonlinear ultrasound simulation on distributed clusters using the k-space pseudospectral method. Bradley E. Treeby, Jiri Jaros (Research School of Engineering, Australian National University, Canberra, ACT 0200, Australia, bradley.treeby@anu.edu.au), Ben T. Cox (Department of Medical Physics and Bioengineering, University College London, London, United Kingdom), and Alistair P. Rendell (Research School of Computer Science, Australian National University, Canberra, ACT, Australia)

Performing realistic simulations of the propagation of nonlinear ultrasound waves through biological tissue is a computationally difficult task. This is because the domain size of interest is often very large compared to the wavelengths of the high-frequency harmonics generated as the ultrasound waves progress. Recently, the k-space pseudospectral method has been applied to this problem to reduce the number of grid points required per wavelength compared to finite difference methods. However, the global nature of the spectral gradient calculations used in this method introduces new challenges for tackling large-scale problems. Here, we discuss three important issues for pseudospectral methods in the context of distributed computing. (1) Decomposing the domain to allow distribution across multiple nodes while still retaining the global accuracy of the spectral gradient calculations. (2) Using non-uniform grids to allow grid points to be clustered around highly nonlinear regions. (3) Avoiding aliasing errors due to modeling nonlinear wave propagation on a fixed grid. For each issue, solutions that retain the efficiency advantages of the pseudospectral method are discussed. We then present recent results of large-scale 3D nonlinear ultrasound simulations in heterogeneous and absorbing media running on both shared memory computers and distributed computer clusters.

8:25

2aBA2. Numerical simulations of three-dimensional nonlinear acoustical waves: Application to the modeling of helicoïdal beams. Régis Marchiano (Institut Jean le Rond d'Alembert (UMR CNRS 7190), University Pierre and Marie Curie, 4 place Jussieu, Paris 75005, France, regis.marchiano@upmc.fr), Jean-Louis Thomas, and Diego Baresch (Institut des NanoSciences de Paris (UMR CNRS 7588), University Pierre and Marie Curie, Paris, France)

A numerical method for the simulation of three-dimensional nonlinear acoustical wave propagation through a homogeneous or weakly heterogeneous medium is presented. This method is based on the resolution a nonlinear wave equation taking into account diffraction, nonlinearities and weak heterogeneities exact up to second order. It is numerically solved in a one-way manner by using a classical splitting method in three steps: angular spectrum method for diffraction, implicit finite differences method for heterogeneities, and semi-analytical Burgers-Hayes method for nonlinearities. Simulations of propagation of nonlinear helicoïdal beams will illustrate the capacities of the method. This kind of acoustical beams featuring radial, azimuthal and axial variations of the field is intrinsically three-dimensional. Furthermore, they have properties of potential interest in biomedical acoustics either for imaging purposes (dynamics of the so-called topological charge) or for therapy purposes (generation of helical shock front or acoustical tweezers with radiation force).

8:45

2aBA3. Rationale behind the Iterative Nonlinear Contrast Source method. Martin D. Verweij, Libertario Demi, and Koen van Dongen (Laboratory of Acoustical Wavefield Imaging, Faculty of Applied Sciences, Delft University of Technology, Delft 2628CJ, Netherlands, m.d.verweij@tudelft.nl)

Modern medical echoscopy increasingly relies on imaging modalities that exploit the features of nonlinear ultrasound. The development of these modalities and corresponding dedicated transducers requires accurate simulations of pulsed nonlinear acoustic wave fields in realistic biomedical situations. This involves nonlinear media with frequency power law attenuation and spatially dependent acoustic properties. Simulations frequently concern strongly steered beams, hundreds of wavelengths long, and their grating lobes. The Iterative Nonlinear Contrast Source (INCS) method is a full-wave method that has been developed for this purpose. It treats the nonlinear term in the Westervelt equation as a contrast source that operates, alongside other source terms, in a homogeneous linear 'background' medium. The background Green's function is known analytically, and convolution with the source terms yields an integral equation. This is solved iteratively to obtain the nonlinear pressure field. The convolution over the four-dimensional computational domain is performed with FFT's, and a grid with only two points per wavelength suffices due to prior filtering of the involved quantities. The present talk elaborates on the characteristic steps of the INCS method, i.e. the contrast source formulation and the filtered convolution. Comparisons with other methods will be made, and recent developments will be presented.

9:05

2aBA4. Numerical schemes for the Iterative Nonlinear Contrast Source method. Koen W.A. van Dongen, Libertario Demi, and Martin D. Verweij (Laboratory of Acoustical Wavefield Imaging, Faculty of Applied Sciences, Delft University of Technology, P.O. Box 5046, Delft 2600 GA, Netherlands, k.w.a.vandongen@tudelft.nl)

Nonlinear acoustics is gaining importance for medical acoustical imaging and high intensity focused ultrasound. With the latter one, high-amplitude acoustic wave fields are used to damage or kill cancer cells. For accurate treatment planning, a full-wave method which can model the propagation and scattering of the nonlinear field in attenuative, heterogeneous media is required. The Iterative Nonlinear Contrast Source (INCS) method is a full-wave method originally developed for homogeneous medium. It recasts the Westervelt equation into an integral equation which can be solved using a Neumann scheme. For heterogeneous media, the same approach results in additional contrast source terms. When these additional contrast sources become strong, convergence of the Neumann scheme may become an issue. To overcome this problem, the Westervelt equation may be linearized and the resulting linear integral equation may be solved with more advanced schemes such as Bi-CGSTAB. Restart strategies may be applied to eliminate systematic errors in the higher harmonics caused by the linearization. However, for realistic wave speed contrasts convergence remains problematic. To overcome these limitations, schemes such as steepest descent may be applied on the original nonlinear integral equation. In the present talk, the different schemes and their pros and cons will be discussed.

9:25

2aBA5. Medical application of nonlinear wave vector frequency domain modeling. Gregory T. Clement (Harvard Medical School, Boston, MA 02115, gclement@hms.harvard.edu) and Yun Jing (Mechanical and Aerospace Engineering, North Carolina State University, Rayleigh, NC)

While the nonlinear properties of tissues have been well documented, the computational demands required to solve nonlinear partial differential equations have generally restricted the dimensionality of numeric studies. These restrictions have been significantly reduced over time, thanks to increased memory and processing availability. Combined with more efficient computational approaches, PC-based three-dimensional modeling of nonlinear fields in tissues is now becoming feasible. Both diagnostic and therapeutic ultrasound could benefit from such accessible modeling, providing a tool for studying customized energy deposition, harmonic signal buildup, parametric methods, transducer characterization, thermometry methods, etc. We will present one such approach, which calculates diffraction through the solution to the homogenous frequency-domain Westervelt equation, while nonlinearity is calculated by the particular solution through a Green's function. The validity and efficiency of the method will be demonstrated by comparison with other well-established methods. This approach also permits backward projection of waves from an initial measurement plane toward the source, allowing a single plane to characterize an entire field, including nonlinear induction of both harmonic and sub-harmonic wave components. This ability will be shown using experimental measurements acquired with a focused source designed for HIFU.

Contributed Papers

9:45

2aBA6. Modeling acousto-optic sensing of high-intensity focused ultrasound lesion formation. Matthew T. Adams, David S. Giraud (Mechanical Engineering, Boston University, 110 Cummington St, Boston, MA 02215, adamsm2@bu.edu), Robin O. Cleveland (Institute of Biomedical Engineering, University of Oxford, Oxford, Oxfordshire, United Kingdom), and Ronald A. Roy (Mechanical Engineering, Boston University, Boston, MA)

Real-time acousto-optic (AO) sensing has been shown to non-invasively detect changes in tissue optical properties, a direct indicator of thermal damage, during high-intensity focused ultrasound (HIFU) therapy. In this work, a comprehensive model is developed to describe the AO sensing of lesion formation during HIFU therapy. The angular spectrum method is used to model ultrasound propagation, and the temperature field due to the absorption of ultrasound is modeled using a finite-difference time-domain (FDTD) solution to the Pennes bioheat equation. Thermal damage dependent optical properties are calculated based on a probabilistic and calibrated thermal dose model. To simulate light propagation inside of insonified and optically heterogeneous tissue, an open-source graphics processing unit (GPU) accelerated Monte Carlo algorithm is used. The Monte Carlo algorithm is modified to account for light-sound interactions, using input from the angular spectrum method, and to account for AO signal detection. Results will show how wavelength and illumination/detection configurations affect the detectability of HIFU lesions using AO sensing.

10:00

2aBA7. Validation of time-domain and frequency-domain calculations of acoustic propagation from breast models derived from magnetic resonance images. Andrew J. Hesford, Jason C. Tillett, Jeffrey P. Astheimer, and Robert C. Waag (Electrical and Computer Engineering, University of Rochester, UR Med Ctr Box 648, Rochester, NY 14642-8648, andrew.hesford@rochester.edu)

Magnetic resonance images with an isotropic resolution of 200 microns were collected for two human breast specimens. The images were interpolated to achieve a resolution of 50 microns and segmented to produce images of skin, fat, muscle, ductal structures, and connective tissues in consultation with a breast pathologist. The images were then mapped to acoustic parameters of sound speed, absorption and density. Calculations of acoustic propagation of fields radiated by point sources outside of the specimens were performed using the k-space finite-difference time-domain method and the frequency-domain fast multipole method. Time-domain k-space results were Fourier transformed and the 3-MHz component was compared to 3-MHz frequency-domain calculations. For the first model, measuring $1180 \times 1190 \times 290$ voxels, the two methods were found to agree in a representative coronal slice to within 5.0% (RMS). The second specimen, comprising $1350 \times 1170 \times 790$ voxels, yielded temporal and frequency-domain results that agreed to within 5.8% (RMS) in a representative coronal slice. Comparable results were obtained in other planes orthogonal to the representative slices. The close agreement establishes confidence in the accuracy of the methods when simulating propagation through large, complicated, realistic models of human tissue.

10:15–10:30 Break

Invited Papers

10:30

2aBA8. High-order numerical methods for nonlinear acoustics: A Fourier Continuation approach. Nathan Albin (Mathematics, Kansas State University, 138 Cardwell Hall, Manhattan, KS 66503, albin@math.ksu.edu)

Dispersion errors, which result from the use of low-order numerical methods in wave-propagation and transport problems, can have a devastating impact on the accuracy of acoustic simulations. These errors are especially problematic in settings containing nonlinear acoustic waves that propagate many times their fundamental wavelength. In these cases, the use of high-order numerical schemes is vital for the accurate and efficient evaluation of the acoustic field. We present a class of high-order time-domain solvers for the treatment of nonlinear acoustic propagation problems. These solvers are based on the Fourier Continuation method, which produces rapidly-converging Fourier series expansions of non-periodic functions (thereby avoiding the Gibbs phenomenon), and are capable of accurately and efficiently simulating nonlinear acoustic fields in large, complex domains.

10:50

2aBA9. Nonlinear modeling as a metrology tool to characterize high intensity focused ultrasound fields. Vera Khokhlova (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, vera@apl.washington.edu), Petr Yuldashev (Physics Faculty, Moscow State University, Moscow, Russian Federation), Wayne Kreider (Applied Physics Laboratory, University of Washington, Seattle, WA), Oleg Sapozhnikov (Physics Faculty, Moscow State University, Moscow, Russian Federation), Michael Bailey, and Lawrence Crum (Applied Physics Laboratory, University of Washington, Seattle, WA)

High intensity focused ultrasound (HIFU) is a rapidly growing medical technology with many clinical applications. The safety and efficacy of these applications require accurate characterization of ultrasound fields produced by HIFU systems. Current nonlinear numerical models based on the KZK and Westervelt wave equations have been shown to serve as quantitatively accurate tools for HIFU metrology. One of the critical parts of the modeling is to set a boundary condition at the source. In previous studies we proposed using measurements of low-amplitude fields to determine the source parameters. In this paper, two approaches of setting the boundary condition are reviewed: The acoustic holography method utilizes two-dimensional scanning of pressure amplitude and phase and numerical back-propagation to the transducer surface. An equivalent source method utilizes one-dimensional pressure measurements on the beam axis and in the focal plane. The dimensions and surface velocity of a uniformly vibrating transducer then are determined to match the one-dimensional measurements in the focal region. Nonlinear simulations are performed for increasing pressure levels at the source for both approaches. Several examples showing the accuracy and capabilities of the proposed methods are presented for typical HIFU transducers with different geometries. [Work supported by NIH EB007643.]

Contributed Papers

11:10

2aBA10. Dual-time-scale method for modeling of the nonlinear amplitude-modulated ultrasound fields. Egor Dontsov and Bojan Guzina (University of Minnesota, 500 Pillsbury Drive SE, Minneapolis, MN 55455, guzina@wave.ce.umn.edu)

This study focuses on modeling of the nonlinear acoustic wave propagation in situations when the amplitude of the focused ultrasound field is modulated by a low-frequency signal. This problem is relevant to both ultrasound imaging applications entailing the use of the acoustic radiation force, and treatment applications such as histotripsy. The difficulty of predicting the pressure wavefield lies in a fact that the excessive length of the low-frequency modulated signal may significantly increase the computational effort. To tackle the problem, this study utilizes the dual-time-scale approach, where two temporal variables are introduced to distinguish between ultrasound-scale and modulation-scale variations. In this case, the Westervelt-type equation can be effectively solved using hybrid time-frequency algorithm for any transient (sufficiently smooth) modulation envelope. To validate the proposed approach, the Khokhlov-Zabolotskaya-Kuznetsov equation was solved in the time domain for an example pressure profile on the boundary. A comparison between the time-domain and hybrid calculations demonstrates that the latter are notably faster, require significantly less memory, and have satisfactory accuracy for the ratios between the modulation and carrier ultrasound frequencies below 0.1.

11:25

2aBA11. Modeling translation of a pulsating spherical bubble between viscoelastic layers. Daniel R. Tengelsen, Todd A. Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Applied Research Laboratories, University of Texas, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

A model was developed previously that enables calculation of the translational force exerted on a pulsating bubble between parallel viscoelastic layers [Hay et al., J. Acoust. Soc. Am. **128**, 2441 (2010)]. Here the translational motion of the bubble is taken into account. Force on the bubble is calculated with a Green's function for the reverberant acoustic field in the channel. The Green's function takes into account not only elastic waves in the channel walls but also viscous boundary layers at the interfaces with the liquid. The dynamical response of the bubble is modeled by an equation of Rayleigh-Plesset form for pulsation, coupled to a momentum equation for translation. The dynamical equations are coupled to the Green's function providing the reverberant pressure field and its gradient acting on the bubble. Calculation of the time-dependent Green's function requires integration over both wavenumber and frequency space at each location along the trajectory of the bubble. Different numerical implementations were considered based on accuracy and efficiency. Simulations will be presented for several combinations of bubble radius, standoff distance, and viscous boundary layer thickness. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and NIH DK070618.]

11:40

2aBA12. Liquid compressibility effects in the dynamics of acoustically coupled bubbles. Derek C. Thomas (Dept. of Physics and Astronomy, Brigham Young University, N283 ESC, Provo, UT 84097, dthomas@byu.edu), Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Applied Research Laboratories, University of Texas at Austin, Austin, TX)

Accurate models for clusters of interacting bubbles are sought for both biomedical and underwater applications. Multiple bubble models have been developed by treating the bubbles as a system of interacting oscillators. The models are obtained initially for bubbles in an incompressible, irrotational, and inviscid liquid; additional effects are included in an *ad hoc* fashion. The existing oscillator models for the dynamics of interacting bubbles are improved by including the effect of liquid compressibility. In particular, while existing models have been improved by including propagation delays in the bubble interactions, the effect of bubble interaction on radiation damping has not been considered. The current work develops corrections for the radiation damping of coupled bubbles in both linear and nonlinear

models of bubble dynamics. These corrections eliminate certain instabilities that have been observed in delay differential equation models of coupled-bubble dynamics. Additionally, an increase in the coupling strength between bubbles undergoing high-amplitude radial motion is predicted when coupled radiation damping is included; this increase in coupling strength strongly affects the predicted motion of the system and the resultant pressure in the surrounding medium. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and NIH DK070618.]

TUESDAY MORNING, 23 OCTOBER 2012 SALON 7 ROOSEVELT, 8:15 A.M. TO 10:00 A.M.

Session 2aEA

Engineering Acoustics: Wideband Transducers and Their Impact on Total Acoustic System Design

Stephen C. Thompson, Chair
Applied Research Lab., Pennsylvania State Univ., State College, PA 16804

Chair's Introduction—8:15

Invited Papers

8:20

2aEA1. Motional current velocity control of piezoelectric loads. Robert C. Randall (Electroacoustic Research Laboratory, Advanced Technology and Manufacturing Center, Fall River, MA 02720, bobrandall81@gmail.com)

It is well known that acoustic interactions affect the transmit radiation pattern of a SONAR array, particularly when the element spacing is small relative to the acoustic wavelength. A negative feedback system with a velocity sense signal fed back to the power amplifier is one method of mitigating the array interactions, and has significant advantages for wideband use compared to either open loop compensation or passive electrical tuning. A velocity control loop flattens the transducer's frequency response, and also reduces the effects of the array interactions proportional to the loop gain. The velocity feedback signal for piezoelectric loads may be obtained by the motional current method, which is equivalent to using an ideal massless accelerometer if the transducer's electrical branch admittance is estimated correctly. The transducer's coupling coefficient, mechanical Q, and a priori estimate of the blocked capacitance fundamentally limits both the maximum stable loop gain, and the output velocity gain and phase tracking relative to the amplifier's input voltage. The array equations governing the acoustical outputs are presented, both with and without motional current velocity control.

8:40

2aEA2. Evaluating transducer bandwidth and effectiveness on overall acoustic system performance. Corey L. Bachand, David A. Brown, and Boris Aronov (BTech Acoustics LLC and UMass Dartmouth, 151 Martine Street, Fall River, MA 02723, corey.bachand@cox.net)

Piezoelectric ceramic cylindrical transducers are used extensively for underwater acoustic communication applications on mobile platforms (UUVs). Employing piezoelectric single crystals, used in either fully-active or active-passive segmented cylinders, in the transducer design has the potential to increase the usable bandwidth while reducing the overall size and weight of the device. In some instances, one single crystal transducer may replace several ceramic transducers and reduce the number of hardware channels. The impact on acoustic system design (power amplifier, matching transformer, and tuning network) for several piezoelectric ceramic and single crystal cylindrical transducer technologies is modeled and supported with measured transducer data.

9:00

2aEA3. A wideband moving coil electrodynamic transducer system for autonomous underwater vehicle-based geoacoustic inversion. Donald P. Massa (Massa Products Corporation, 280 Lincoln St., Hingham, MA 02043, Massa@massa.com) and Juan I. Arvelo (Applied Physics Laboratory, Johns Hopkins University, Laurel, MD)

A small expendable wideband low-frequency sound source that will be deployed on the seafloor is being developed to be used for geoacoustic inversion surveys in conjunction with a terrain-hugging AUV. This low-cost deployable source contains a transducer that produces a relatively flat transmit response over the broad frequency band of 100 to 4000 Hz. In operation, a seafloor interface wave will be excited and exploited for geoacoustic inversion by the deployed sound source and a receiving array on the bottom-hugging AUV. A feasibility study is also being performed that includes physics-based sonar simulations to infer the performance of geoacoustic inversion in a number of AUV scenarios and environmental conditions. Based on this study, design trade-offs will be determined to finalize key factors of the transducer, such as its physical size, weight, and production cost. Battery technology is also being developed to optimize the source level, the duty cycle, and the operating life of the signals that will be transmitted during data collection. This effort is being supported by ONR.

9:20

2aEA4. Array synthesis and wide band system. Dehua Huang (NAVSEANPT, Howell St., Newport, RI 02841, DHHuang@cox.net)

Acoustic arrays play important role as the key devices in wide band system designs. The Laguerre polynomials are successfully applied to the array synthesis first time. Conventional uniform array is the simplest design, because each element is excited at the same weight, which leads to high side lobe levels, artifacts and noise to advanced systems. Dolph utilized the first kind Chebyshev polynomials to synthesize the array beam pattern for side lobe control. However, it comes with the equal levels of the side lobes, due to the mathematical nature of the first kind Chebyshev polynomials. Taylor introduced a modified version Dolph-Chebyshev synthesis technique, which displayed tapered down side lobe levels in the region away from the main lobe. The key characteristics from the sample numerically simulated arrays by the Laguerre polynomials synthesis, i. e. radiation patterns, half-power beam width, directivity and the beam efficiency are compared with those from the synthesis of the Dolph-Chebyshev of the first kind or second kind, Taylor shading, Legendre and Hermite polynomials techniques. Work supported by the U.S. Navy.

9:40

2aEA5. High intensity air ultrasound source for determining ultrasound microphone sensitivity up to 400 kHz. Angelo J. Campanella (Acculab, Campanella Associates, 3201 Ridgewood Drive, Ohio, Hilliard, OH 43026, a.campanella@att.net)

A broadband air jet ultrasound source, RSS101U-H, for animal bioacoustics research produces broadband ultrasound to at least 400 kHz. Free field reciprocity calibration of 1/2" condenser microphones on-axis, grid caps removed, using sine wave excitation by the method of Rudnick and Stein [JASA 20, pp 818-825, (1948)] was made (previously used to 280 kHz [JASA 67, p 7, (1980)]). Measurement from 5 kHz to 100 kHz was made in 1 kHz bins via an FFT analyzer. A communications receiver was used from 40 kHz to 400 kHz. The sensitivity of a 1/4" microphone was determined from the free field of the reciprocity 1/2" microphone source. Air jet ultrasound level at 80 mm distance was then determined with the 1/4" microphone. Communications receiver 2.5 kHz bandwidth data was reduced to 1 kHz bin values. Air humidity sound absorption was determined via ANSI 1.26. The 1/4" microphone sensitivity and broadband source sound level results in 1 kHz bands to 400 kHz are presented. Air jet spectral level was 97 dB re 20 uPa @ 75 kHz to 57 dB @ 400 kHz. This can be used to rapidly determine the sensitivity of any air ultrasound microphone over this frequency range.

TUESDAY MORNING, 23 OCTOBER 2012 ANDY KIRK A/B, 7:55 A.M. TO 12:00 NOON

Session 2aED

Education in Acoustics: Engaging and Effective Teaching Methods in Acoustics

Wendy K. Adams, Cochair
Physics, University of Northern Colorado, Greeley, CO 80631

Preston S. Wilson, Cochair
Applied Research Lab., Univ. of Texas at Austin, Austin, TX 78712-0292

Chair's Introduction—7:55

Invited Papers

8:00

2aED1. Collaborating to improve science teaching and learning through the ComPADRE digital library I. Bruce Mason (Physics & Astronomy, University of Oklahoma, Norman, OK 73019, bmason@ou.edu) and Lyle Barbato (AAPT, College Park, MD)

Most educators have found that improving their classes is best done as a collaborative process, by sharing best practices and resources with others. The ComPADRE digital library has been supporting these collaborations for the past decade through a vetted, online database of teaching and learning materials, personalization services, and tools for groups to interact. This talk will explore some examples of the resources available through the ComPADRE database that can be used to engage students in learning and can help instructors improve the outcomes of their courses. It will cover the organization of materials and how ComPADRE members can meet their personal needs. The talk will also explore examples of ComPADRE collections built by and for communities of teachers interested in specific topics or courses. Of course, examples of fun and engaging learning materials will also be demonstrated. ComPADRE is a collaboration of the American Association of Physics Teachers, the American Institute of Physics, the American Physical Society, and the Society of Physics Students and is part of the National STEM Digital Library. It is supported, in part, by funding of the National Science Foundation.

8:40

2aED2. Teaching musical acoustics with clickers. William Hartmann (Physics-Astronomy, Michigan State University, 4208 BPS Bldg., East Lansing, MI 48824, hartman2@msu.edu)

Musical acoustics is a well-proved avenue for teaching scientific concepts to students whose fields of study and interests are far removed from any science. In recent years the Michigan State course in musical acoustics has benefited greatly from using clickers. Frequent clicker questions (1 point for any answer; 2 points for the correct answer) promote attendance and help maintain a lively, interactive classroom environment, even for a large lecture class. Nobody sleeps when clicker points are on the line. Students are encouraged to discuss responses to clicker questions among themselves before answering, and the response protocol allows students to change their responses at any time before the polling is closed. Musical acoustics lectures include many demonstrations that can be presented as experiments requiring students to predict the result in advance using their clickers. "No-count" or "all-good" clicker questions can be used to determine student responses to perceptual experiments, and the feedback from the scoring algorithm gives the answer and the inevitable variability. Most important, responses to clicker questions give an instructor instant feedback about whether new lecture material has been understood. To use clickers in this way requires flexible instruction and spontaneous generation of new clicker questions.

9:00

2aED3. Providing interactive engagement in introductory acoustics through design-intensive laboratories. Andrew Morrison (Natural Science Department, Joliet Junior College, Joliet, IL 60431, amorrison@jjc.edu)

More than three decades worth of education research has shown with overwhelming evidence that the best way for students to learn is to be actively engaged in the classroom rather than passively taking in material delivered from an instructor. Although the reform of introductory classes has been widely adopted by many instructors, the implementation of reformed introductory laboratory curricula has not been as widely adopted. In our introductory acoustics course, students complete design-intensive labs where much of the instruction has been stripped away. The emphasis on student-driven experiment design and analysis is intended to provide a more scientifically authentic experience for students. The course is taught using an integrated lecture and laboratory approach. An overview of the laboratory framework and example laboratory activities used in our introductory acoustics class will be presented.

9:20

2aED4. Techniques for teaching building acoustics and noise control to university architecture students. Robert C. Coffeen (School of Architecture, Design & Planning, Universtiy of Kansas, 1465 Jayhawk Blvd, Lawrence, KS 66045, coffeen@ku.edu)

Architects are visual people. And, we cannot see sound in an architectural venue. Perhaps this has something to do with their historically poor record in dealing with acoustic and noise control issues in building spaces. Experience in teaching architecture students indicates useful teaching techniques include visits to venues with both suitable and unsuitable acoustic conditions, using modeling and auralization so that students can hear simulations of acoustical conditions produced by various interior surface shapes and architectural materials, relating their actual listening experiences in venues of various types to interior surface shapes and finish materials, and discussing the acoustical characteristics of interior materials so that a visual inspection of a space can lead to a general determination of the room acoustic conditions to be anticipated. Also discussed will be techniques for teaching architecture students the basics of architectural noise control and the basics of mechanical system noise control.

9:40

2aED5. Acoustic tweets and blogs: Using social media in an undergraduate acoustics course. Lily M. Wang (Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

Each fall, the author teaches an undergraduate architectural acoustics course to around 40 third-year architectural engineering students at the University of Nebraska. Beginning in 2011, a social media component was introduced to explore the use of this technology and how it may supplement the students' learning experience. Students were given an opportunity to receive extra credit by using twitter and/or blogging about course material using a set hashtag (#AE3300) or through the course website. Results were positive, and the author will discuss pros and cons that she has experienced in adding this social media component. Suggestions for future implementations and examples of student participation will be presented.

10:00–10:15 Break

10:15

2aED6. Use of pre-class quizzes to promote active learning in acoustics. Kent L. Gee and Tracianne B. Neilsen (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT 84601, kentgee@physics.byu.edu)

Classroom instruction can be inefficient or ineffective when students do not come to class prepared. One strategy to engage students prior to class is the use of pre-class quizzes. One pedagogical method developed for introductory courses by physics education researchers is pre-class "just-in-time-teaching" quizzes. As a variation on that idea, pre-class learning activities have been used with great success in the general education acoustics course at Brigham Young University (BYU). However, such methods are not often applied at the advanced undergraduate and graduate levels. This paper reviews some of the findings from the introductory course efforts and then describes the implementation of pre-class quizzes for two advanced acoustics courses at BYU. Two lessons learned thus far are 1) the questions, which have a free-response format, must be carefully constructed so that the instructor can gauge student understanding, and 2) when successfully implemented, the quizzes can provide an effective framework for a class discussion of a topic, rather than a lecture with little to no participation.

10:35

2aED7. Active-learning techniques in an introductory acoustics class. Tracianne B. Neilsen and Kent L. Gee (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT 84602, tbn@byu.edu)

The goal of active-learning techniques is to encourage the students to become involved with the material and take ownership for their learning, which fosters long-term knowledge and enjoyment of the subject. In this era of student-based learning outcomes, an active-learning approach is important because it focuses on what the students are doing to facilitate learning instead of what the instructor is trying to teach. To benefit most from class time, the students need to have the opportunity to actively engage with the material beforehand. If meaningful pre-class activities are required, it is easier to interact with the students during class. Some key methods for encouraging active learning during class include incorporating their pre-class experiences, conducting discussions, encouraging student participation, and evaluating student understanding with a response system, such as i-clickers. After the class time, students need apply what they have learned in answering additional questions on homework assignments and in hands-on laboratory experiences. Lessons learned after several years' worth of step-by-step efforts to approach these goals in an introductory acoustics class, which serves a wide range of majors as a general science elective, are presented.

10:55

2aED8. 25 years of distance education in acoustics. Daniel A. Russell and Victor W. Sparrow (Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802, drussell@engr.psu.edu)

Online education is quickly becoming a popular means of delivering course content to the masses. Twenty-five years ago this Fall, the Graduate Program in Acoustics at Penn State began offering graduate level instruction in acoustics to students at a distance. In 1987 courses were offered via satellite links to Navy and industry labs, with PictureTel video conferencing to two centralized locations added in 1992. By 1994 videotapes allowed for broader content distribution to students at more varied locations. In 2002 videostreaming lectures over the internet expanded the delivery even further. Currently courses are taught to a blended audience of both resident and distance students, with lectures being live streamed over the internet and archived digitally for offline access. This talk will briefly summarize the history and development of online graduate education in acoustics at Penn State. We will discuss the use of technology both as a tool for delivering course content as well as the impact that technology has on the quality and means of instruction and the interaction between teacher and students. The necessary adaptation of teaching styles and the adjustments required to meet the varied needs of a blended student audience will also discussed.

Contributed Papers

11:15

2aED9. Problem solving assessment. Wendy K. Adams (Physics, University of Northern Colorado, CB 127, Greeley, CO 80631, wendy.adams@colorado.edu)

Although educators and employers highly value problem solving and have put extensive effort into understanding successful problem solving, there is currently no efficient way to evaluate it. Science educators regularly make use of concept inventories and perceptions surveys (aka: attitudes and beliefs) to evaluate instruction. However, these only touch on a fraction of what is learned in a course. Students apply a range of processes, expectations and bits of knowledge when solving a physics problem and some of these are impacted by the course. The question is how can we identify what these processes, expectations and knowledge are, how can we teach them and then how can we measure them? While developing the CAPS (Colorado Assessment of Problem Solving), I identified 44 processes, expectations and bits of knowledge used to solve an in depth real world problem. In this presentation CAPS and some of what was learned during the development will be presented.

11:30

2aED10. Teaching graduate level acoustics courses to a blended enrollment of resident and distance education students. Daniel A. Russell (Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, PA 16802, drussell@engr.psu.edu)

The Graduate Program in Acoustics offers courses leading to the M.Eng. in Acoustics online through Penn State's World Campus. The current method of course content delivery is to live-stream (with digital video archive) lectures to a blended student audience consisting of about 10-15 resident students physically present in the classroom and another 15-20 students at a distance who may be watching the class live, or who may be viewing the archived recording afterward. This paper will explore several issues involving the engagement of this blended audience of students. How does one encourage and enable students with a broad range of backgrounds, interests, and physical locations to engage with the topic material? How does one foster collaboration and interaction between distance students and the teacher and between resident and distance students? How does one manage office hours, help sessions, group projects, experiments, and student presentations for a blended student audience? Current practice and personal experiences from our faculty will be shared, and ideas from the audience will be welcomed.

11:45

2aED11. Real-time audio signal capture and processing using MATLAB object oriented programming. Samarth Hosakere Shivaswamy, Xiang Zhou, Stephen Roessner, Gang Ren (Dept. of Electrical and Computer Engineering, Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu), Dave Headlam, and Mark Bocko (Dept. of Electrical and Computer Engineering; Dept. of Music Theory, Eastman Sch. of Music, Univ. of Rochester, Rochester, NY)

In MATLAB programming language the real-time audio processing functions are usually simulated in non-real-time due to a lack of real-time audio programming support. As a result the real-time audio signal capture and processing functionalities are usually implemented in other programming languages and cannot utilize the extensive signal processing functionalities provided by MATLAB. In this paper we introduce a MATLAB real-time signal processing framework based on MATLAB timer object and audiorecorder object. The proposed processing framework serves as an alternative solution for real-time programming implementation and demonstration. In our proposed processing framework the timer object is implemented to handle the looping of processing cycle, schedule the signal processing tasks, and handle the error processing. The audio capturing/processing functionality is implemented in the timer cycle by using two audiorecorder objects that read the audio streaming data and feed a segment of data to signal processing alternatively. The proposed framework achieves satisfactory real-time performance with no missing audio frames when a short audio delay setting of 10ms is applied. Several application examples of our proposed framework are also demonstrated.

Session 2aNS

Noise and Architectural Acoustics: Sound Quality, Sound Design, and Soundscape

Brigitte Schulte-Fortkamp, Cochair
TU Berlin, Einsteinufer 25 TA, Berlin 10587, Germany

Klaus Genuit, Cochair
HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Sq., Vernon, CT 06066

Chair's Introduction—7:45

Invited Papers

7:50

2aNS1. Approaching environmental resources through soundscape. Brigitte Schulte-Fortkamp (TU Berlin, Einsteinufer 25 TA 7, Berlin 10587, Germany, bschulte_f@web.de)

The Soundscape concept is introduced as a scope to rethink the evaluation of "noise" and its effects and to focus on a diverse field of experts and expertise in order to fulfill the requirements for a "good environment" or a "sensitive environment" with respect to quality of life. Moreover, Soundscape is defined as an environment of sound with emphasis on the way it is perceived and understood by the individual, or by a society. Therefore it is suggested to explore noise in its complexity and its ambivalence and its approach towards sound to consider the conditions and purposes of its production, perception, and evaluation, to understand evaluation of noise/ sound as a holistic approach. Qualitative methods referring to a heterogeneous 'field of research' and among them are different forms of observation, interviewing techniques and the collection of documents or archival data as well as binaural measurements will be presented and proven regarding their effects of explanations. The intention of scientific research here is to learn about the meaning of the noise with respect to people's living situation and to implement the adequate procedure to open the "black box" of people's mind regarding their needs for a supportive environment.

8:10

2aNS2. Relationship between environmental noise, sound quality, soundscape. Klaus Genuit, André Fiebig (HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de), and Brigitte Schulte-Fortkamp (Technical University Berlin, Berlin, Germany)

The term "Environmental Noise" is well known for many years. Its characteristic is often described by parameters like A-weighted SPL, Lden, Lday, Levening, Lnight. These parameters can be measured and calculated. In the field of "Sound Quality" psychoacoustic parameters are additionally used like loudness, sharpness, roughness and others, which can be measured but not calculated for a complex sound field. On an international level a standard is available only for the loudness of stationary sounds so far. The "relatively" young term "soundscape" will be standardized in ISO 12913-1. Moreover, as it considers human perception including cognitive aspects, context and interaction it goes beyond physics and psychoacoustics. It involves a concept, where environmental noise is not reduced to an averaged quantity evoking only unpleasantness feelings estimated by statistical probabilities, but understanding noise as a valuable resource, which can be purposefully utilized. In spite of recent progresses in the standardization process lots of misinterpretations occur in practical use, where the terms are heavily mixed up. Environmental noise and soundscape are no synonyms, for example low noise level does not directly mean a good sound quality. The paper will clarify options and limitation of both terms.

8:30

2aNS3. Soundscape and sound quality—Similar and powerful design techniques. Bennett M. Brooks (Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com)

Two powerful analysis techniques available to acoustical researchers and designers include the sound quality method and the very similar soundscape method. In each of these techniques physical acoustical measurements are combined with in-depth interviews and opinion juries to determine the cause and effect relationship that a particular sound, or set of sounds, has on a population. The sound quality technique has been in use for many years, and focuses on product development. An example is the sound of an automobile door closing - is the car door closing sound perceived as "solid and expensive" or "cheap and tinny". Another example is a vacuum cleaner - does it sound "powerful and effective" or "weak and ineffective"? The soundscape technique focuses on environmental sound, often in public spaces like a park or in a residential neighborhood. For example, is a certain transportation vehicle sound or outdoor entertainment facility sound acceptable or unacceptable to the wider community? This paper will explore the similarities between these two related fields and the opportunities they offer to sound designers.

8:50

2aNS4. The sound-absorbing city—New ideas for living environments around airports. Juergen Bauer (Department of Architecture, Waterford Institute of Technology, Granary, Hanover St, Co Waterford, Ireland, jbauer@wit.ie)

Great efforts and progress have been made in terms of noise protection measures in both urban, suburban and rural environments. Local or regional urban planning guidelines and anti-noise-manuals provide experienced and practical advice to reduce noise, in order to provide a better quality of life. Most of the anticipated solutions such as noise-protection-walls, fences, planted mounds etc. will address issues caused by land traffic. However, due to their nature, they fail to respond to "airborne" noise immission. In addition, there is a common public misconception that sound should be interpreted as noise ie. as a waste to get rid of, instead of critically identifying sound and sound clusters as a potential and as a resource to be integrated. The concept of the sound-absorbing city applies the same principles of sound reflection and sound absorption, as applied to an architectural space, in an urban space. It also investigates the possibility of combining unwanted sound, such as air traffic noise with wanted sound, such as nature and community sound. The paper discusses the concept of the sound-absorbing city, its potentials and its apparent limits, with regard to new settlements and existing agglomerations around airports.

9:10

2aNS5. I hear what you mean—Source intensity versus receiver level. Alex Case (Sound Recording Technology, University of Massachusetts Lowell, Lowell, MA 01854, alex@fermata.biz)

A receiver's assessment of the quality of any single element of a soundscape is not limited to the objective values of sound attributes at the receiver location, but includes an intuitive or instinctive compensation for the source-to-receiver path. Borrowing from the time-proven connection between performance intensity versus presented level in sound recordings, emotion and meaning are found to include what a listener infers about the local environment and context at the sound source, discounting whatever may have happened on its way to the listener.

2a TUE. AM

9:30

2aNS6. Relevance and applicability of the soundscape concept to physiological or behavioral effects caused by noise at very low frequencies which may not be audible. Wade Bray (HEAD acoustics, Inc., 6964 Kensington Road, Brighton, MI 48116, wbray@head-acoustics.com)

A central tenet of the Soundscape concept is that humans immersed in sonic environments are objective measuring instruments (New Experts), whose reports and descriptions must be taken seriously and quantified by technical measurements. A topic category in acoustics meetings of recent years is "Perception and Effects of Noise." There is growing evidence from the field, and from medical research, that the ear's two-part transducer activity involving inner hair cells (IHC, hearing, velocity-sensitive) and outer hair cells (OHC, displacement-sensitive) may, through demonstrated OHC activation and neural signals at up to 40 dB below the audibility threshold, produce behavioral and physiological effects as reported by a growing number of people. The Soundscape concept centering on human responses, New Experts, is as important and applicable to responses to effects from sound as it is to responses to directly audible sound. In a wider sense, this is a new sound quality and psychoacoustic issue.

9:50

2aNS7. Soundwalks in urban areas—Triangulation of perceptive and acoustical data. Kay S. Voigt and Brigitte Schulte-Fortkamp (Institute of Fluid Mechanics and Engineering Acoustic, Technische Universitaet Berlin, Einsteinufer 25, Berlin 10587, Germany, kay.s.voigt@gmx.de)

Current work takes a comparative view on the analysis of several soundwalks in urban areas, investigating the benefit of additional effort in enhancing the attendees' description-level for data triangulation. Soundwalk is a tool in the soundscape approach for an all-embracing analysis of the unique sonic environment. The triangulation of data has to combine acoustical measurements of the same procedures and perceptive appraisals differing in its quality of description. The focus of each research varies from perception of places to questions of the overall feeling of safety at the different locations. Local, acoustical and safety experts are involved. The qualitative analysis considers several variables like profession of soundwalkers, knowledge about places, kind of places, chosen or given locations, and the used native language of questioning. Different levels of description narrated by the participators will be identified, as well as its possible emphasis by discourse on attendees' scaled ratings and written notes. This analysis progress contributes to an appropriate assignment of subjective descriptions to values of psychoacoustical parameters and the elucidation of predominant aspects in the soundscape. Furthermore the soundwalk's contrasting capacity with regards to the content of previous interviews detected multiple layers on the issue of safety in municipal locations.

10:10–10:25 Break

10:25

2aNS8. Sound preference prediction in a design stage—A case study in the Shenzhen Dongmen Open Space. Lei Yu (HIT Shenzhen Graduate School, E425 HIT Campus, Shenzhen University Town, Xi Li, Nan Shan, Shenzhen 518055, China, Leilayu@hitsz.edu.cn)

Attention on visual effects is insufficient in urban open spaces, while soundscape is complementation and sound design is crucial. In this paper, sound effects in the Shenzhen Dongmen Open Space have been studied. It shows that exist sounds are not delightful to satisfy acoustic comfort but to cause annoy perceptions. Therefore, this study is focused on examining various sound effects in the Shenzhen Dongmen Open Space concerning on sound preference evaluations. Based on various sounds influencing the subjective preference evaluations, Artificial Neural Network (ANN) models have been developed to predict how delightful of the sonic environment in terms of different sound design schemes to the space. Furthermore, the sound preference predictions of ANN models' output will be compared with the preference evaluations from lab experiments, and then validated by the lab results.

Contributed Papers

10:45

2aNS9. Soundscape analysis of two parks in Berlin. Natalia Manrique-Ortiz and Brigitte Schulte-Fortkamp (Technische Universität Berlin, Einsteinufer 25, Berlin 10587, Germany, natalia.manrique.ortiz@gmail.com)

Nowadays the protection of quiet areas is an issue of increasing importance. This importance is reflected in the European directives and policy intentions of many countries around the world. In order to protect this areas, is important to characterize their soundscapes and analyze the areas, paying special attention to the geography, aesthetics, social, psychological and cultural aspects, since these aspects play a significant role in the noise perception. The aim of this research is to analyze two of the most important Berliner parks. Victoria-Park and Schlosspark are located in very different areas in Berlin. Schlosspark is in a more quiet neighborhood with families and Victoria-Park is in a lively, young and multicultural neighborhood. The protection of these parks begins understanding the community who make use of them. The research will be based on interviews and soundwalks according to the soundscape approach. The results will be presented.

11:00

2aNS10. Sound and noise in urban parks. Antonio P. Carvalho (Laboratory of Acoustics, University of Porto, FEUP - Fac. Eng. Univ. Porto, DEC (NIF501413197), Porto P-4200-465, Portugal, carvalho@fe.up.pt) and Ricardo C. Dias (Laboratory of Acoustics, University of Porto, Porto, Portugal)

The main goal of this work is to study the soundscape of urban gardens and parks using a sample of ten sites in Porto, Portugal to characterize their noise levels through the acoustic characterization of the park's exterior and interior noise levels (LAeq, LA10, LA50 and LA90) and by a socio-acoustic survey to the visitors to check their perception of acoustic quality. The measurements showed gardens/parks with interior noise levels from 47 to 61 dBA (with maximum exterior noise levels up to 67 dBA). The difference between exterior and interior LAeq was between 3 and 19 dBA. The gardens with lower noise levels are the larger and out of downtown. An "acoustic" classification for gardens/urban parks is proposed regarding their noise "isolation" capacity and acoustic ambience. Measurements done in 1990 allow for the comparison of the evolution in the last 21 years. The socio-acoustic survey concludes that Porto's city parks are visited mostly by an elderly male population that regards these places as sites of gathering and to practice some physical activity rather than as an acoustic retreat. The population seems accustomed to the dominant noise, classifying these spaces as pleasant and quiet, even when noise is over acceptable limits.

11:15

2aNS11. Investigation of tranquility in urban religious places. Inhwan Hwang, Jooyoung Hong, and Jin Yong Jeon (Architectural Engineering, Hanyang University, Seoul, Seongdong-gu 133791, Republic of Korea, jyjeon@hanyang.ac.kr)

In the present study, tranquility in urban religious places including a cathedral and a Buddhist temple has been assessed by soundwalks. Both Myung-dong Cathedral and Bongeun Temple located in the center of Seoul were selected as measurement sites. During the soundwalks, audio-visual recordings were conducted at selected positions. From the field measurements, the temporal and frequency characteristics of the sound environment in two religious places were explored. Participants evaluated their perceived soundscape using a soundwalk questionnaire along the soundwalk routes in the church and temple gardens in order to investigate the value of tranquility as urban stress relievers. From the results, indicators representing tranquility difference in particular soundscapes were examined.

11:30

2aNS12. Psychoacoustic assessment of a new aircraft engine fan noise synthesis method. Selen Okcu (National Institute of Aerospace, Hampton, VA 23666, selen.okcu@nasa.gov), Matthew P. Allen (Department of Mechanical Engineering, Virginia Tech, Blacksburg, VA), and Stephen A. Rizzi (Structural Acoustics Branch, NASA Langley Research Center, Hampton, VA)

Simulation of aircraft flyover events can facilitate psychoacoustic studies exploring the effects of noise generated by future aircraft designs. The perceived realism of a simulated flyover event may be impacted by the perceived realism of the synthesized fan noise of the aircraft engine. Short-term fluctuations in tonal amplitude and frequency are important cues contributing to that perception of realism, but are not accounted for by predictions based on long-term averages. A new synthesis method has been developed at NASA Langley Research Center to generate realistic aircraft engine fan noise using predicted source noise directivities in combination with short-term fluctuations. In the new method, fluctuations in amplitude and frequency are included based upon analysis of static engine test data. Through psychoacoustic testing, this study assessed perceived effectiveness of the new synthesis method in generating realistic fan noise source. Realism was indirectly assessed by judging the similarity of synthesized sounds (with and without fluctuations) with recordings of fan noise. Results of ANOVA analyses indicated that subjects judged synthesized fan noise with fluctuations as being more similar to recordings than synthesized fan noise without fluctuations.

11:45

2aNS13. A geospatial model of ambient sound pressure levels in the continental United States. Dan Mennitt, Kurt M. Fristrup (Natural Sounds and Night Skies Division, National Park Service, Fort Collins, CO 80525, daniel_mennitt@partner.nps.gov), and Kirk Sherrill (Inventory and Monitoring, National Park Service, Fort Collins, CO)

There has been much effort in the US and worldwide to measure, understand and manage natural soundscapes which are often complex due to a multitude of biological, geophysical, and anthropogenic influences. The sound pressure level is a time and space varying quantity that represents the aggregate of present sources. This work presents a predictive model relating seasonal sound pressure levels to geospatial features such as topography, climate, hydrology and anthropogenic activity. The model utilizes random forest, a tree based machine learning algorithm, which does not explicitly incorporate any apriori knowledge of acoustic propagation mechanics. The response data encompasses 271,979 hours of acoustical measurements from 192 unique sites located in National Parks across the contiguous United States. Cross validation procedures were used to evaluate model performance and identify GIS explanatory variables with predictive power. Using the model, the effect of individual explanatory variables on sound pressure level can be isolated and quantified revealing trends across environmental gradients. An example application of projecting predicted sound pressure levels across the Olympic peninsula is discussed. Because many wildlife habitats, geological processes, and anthropogenic impacts occur on a regional scale, the extent of acoustical analyses must be on similar scales.

Session 2aPA

Physical Acoustics: Waves in Heterogeneous Solids I

Joseph A. Turner, Cochair
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Goutam Ghoshal, Cochair
Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801

Chair's Introduction—7:55

Invited Papers

8:00

2aPA1. Ultrasound therapy delivery and monitoring through intact skull. Kullervo Hynynen (Medical Biophysics, University of Toronto, Sunnybrook Health Sciences Centre, Toronto, ON M4N 3M5, Canada, khynynen@sri.utoronto.ca), Yuexi Huang, Meaghan O'Reilly (Physic Sciences Platform, Sunnybrook Research Institute, Toronto, ON, Canada), Ryan Jones, Dan Pajek (Medical Biophysics, University of Toronto, Toronto, ON, Canada), and Aki Pulkkinen (Physic Sciences Platform, Sunnybrook Research Institute, Toronto, ON, Canada)

Magnetic Resonance imaging guided and monitored focused ultrasound is now tested for deep focal thermal ablation of brain tissue in clinical setting. The major barrier for this treatment is the propagation of ultrasound through an intact skull that strongly attenuates and scatters the ultrasound wave. The distortion can be corrected by using CT-derived bone density information and computer simulations to derive phase and amplitude information such that the driving signals can be adjusted to reduce the distortions. In this paper the current results on ultrasound propagation through skull will be discussed and the clinical applications of noninvasive ultrasound treatments will be reviewed. The advance in online acoustic monitoring of cavitation based treatments methods will also be shown

8:20

2aPA2. Modeling of transcranial ultrasound for therapeutic and diagnostic applications. Gregory T. Clement (Harvard Medical School, Boston, MA 02115, gclement@hms.harvard.edu)

Ultrasound's use in the brain has conventionally been limited by its inability to penetrate the skull. To overcome these limits, we have been investigating techniques to maximize energy transfer and minimize distortion through the skull bone. These model-based aberration correction approaches - now in the early stages of clinical testing - rely on both practical and accurate numeric methods. Efforts to improve these methods necessitate an increasingly detailed consideration of skull heterogeneity. To facilitate this numerically-intensive problem, we are utilizing an inhomogeneous pressure simulation code, based on a pseudo-spectral solution of the linearized wave equation. Forward and scattered waves are determined over a pre-specified volume with scattering determined by the impedance mismatch between a given voxel and regional points in the projection plane. The total forward-scattered pressure is recorded over the relevant k-space, while reflected energy is processed in a separate backward projection. This process is repeated iteratively along the forward projection plane until the volume of interest has been traversed. This procedure can be repeated an arbitrary number of times N, representing N-1 order scattering. Abilities and limitations of the method will be demonstrated by comparison with FDTD simulation.

8:40

2aPA3. Validation of a finite-difference acoustic propagation model of transcranial ultrasound. Guillaume Bouchoux, Kenneth B. Bader (Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, bouchoge@ucmail.uc.edu), Joseph J. Korfhagen (Neuroscience Graduate Program, University of Cincinnati, Cincinnati, OH), Jason L. Raymond (Biomedical Engineering Program, University of Cincinnati, Cincinnati, OH), Shivashankar Ravishankar, Todd A. Abruzzo (Radiology, University of Cincinnati, Cincinnati, OH), and Christy K. Holland (Internal Medicine, University of Cincinnati, Cincinnati, OH)

Adjuvant ultrasound exposure improves rtPA thrombolysis in stroke patients. Transmission of 120-kHz ultrasound through the temporal bone is efficient but exhibits skull-dependent distortion and reflection. Numerical models of acoustic propagation through human skull based on computed tomography (CT) data have been developed. The objective of our study was to validate a finite-difference model of transcranial ultrasound quantitatively. The acoustic fields from a two-element annular array (120 kHz and 60 kHz) were acquired in four ex-vivo human skulls with a calibrated hydrophone (10 kHz-800 kHz frequency range). The spatial distributions of the acoustomechanical properties of each skull were obtained from CT scans and used for simulations. Predicted acoustic fields and waveform shapes were compared with corresponding hydrophone measurements and were in good agreement. Transmitted wave amplitudes were systematically underestimated (14%) and reflected wave amplitudes were overestimated (30%). The acoustic impedance of each skull was likely underestimated from the CT scans. However, high correlation between predictions and measurements (R<+>2<+>=0.93 and R<+>2<+>=0.88 for transmitted and reflected wave amplitudes, respectively) demonstrates that this model can be used quantitatively for evaluation of 120-kHz ultrasound-enhanced thrombolysis. This work was supported by NIH-RO1-NS047603.

9:00

2aPA4. Ultrasound assessment of bone with ultrasound: Present and future. Pascal Laugier (Laboratoire d'Imagerie Parametrique, CNRS/University Pierre et Marie Curie, 15 rue de l'ecole de medecine, Paris 75017, France, pascal.laugier@upmc.fr)

Bone is a composite, porous and anisotropic material whose complex hierarchical structure extends over several levels of organization from the nanoscale to the macroscopic scale. One of the striking features of this tissue is its ability to adapt to variable loading conditions. This results in spatially, temporally and directionally variable elastic properties leading to a perfect adaptation to locally varying functional demands. Elastic properties of bone are nowadays widely used in fundamental studies, in conjunction with numerical models, to investigate the structure-function relationships and in clinical applications to predict fracture risk or to monitor fracture healing. However, the problem of multiscale assessment of bone elastic properties, spanning the full range of applications from in vitro to in vivo applications, remains a challenge. Novel emerging quantitative ultrasound technologies, taking benefit of the scalability of ultrasound, have emerged to noninvasively investigate elastic properties at multiple organization level. These include scanning acoustic microscopy, ultrasonic resonant spectroscopy and guided waves propagation. These techniques will be presented to show how they can help in characterizing the anisotropic stiffness tensor in vitro or determine bone properties in vivo. Relationships of quantitative ultrasound variables with structural and elastic alterations will be illustrated through multiple examples.

Contributed Papers

9:20

2aPA5. Plumbing the depths of Ligeia: Considerations for acoustic depth sounding in Titan's hydrocarbon seas. Juan I. Arvelo and Ralph Lorenz (Applied Physics Laboratory, Johns Hopkins University, 11100 Johns Hopkins Rd., Laurel, MD 20723, juan.arvelo@jhuapl.edu)

Saturn's moon Titan is the only satellite in our solar system with a dense atmosphere and hydrocarbon seas. The proposed Titan Mare Explorer (TiME) mission would splashdown a capsule to float for 3 months on Ligeia Mare, a several-hundred-kilometer wide sea near Titan's north pole. Among TiME's scientific goals is the determination of the depth of Ligeia, to be achieved with an acoustic depth-sounder. Since Titan's surface temperature is known to vary around 92 K, all instruments must be ruggedized to operate at cryogenic temperatures. This paper's contributions include an approach to infer key acoustic properties of this remote environment, their influence on the development of a cryogenic depth sounder, and on an approach to infer the transducer's response, sensitivity and performance when unable to perform in-situ calibration measurements or to replicate key environmental conditions. This effort was conducted under the auspices of the Civilian Space Independent Research and Development program from the Johns Hopkins University Applied Physics Laboratory.

9:35

2aPA6. Acoustophoresis in gases: Effect of turbulence and geometrical parameters on separation efficiency. Etienne Robert, Ramin Imani Jajarmi (Mechanics, Kungliga Tekniska Högskolan (KTH), Osquars Backe 18, Stockholm 100 44, Sweden, etienne@mech.kth.se), Markus Steibel, and Klas Engvall (Chemical Technology, Kungliga Tekniska Högskolan (KTH), Stockholm, Stockholm, Sweden)

Advanced particle manipulation techniques based on acoustophoresis have been developed in recent years, driven by biomedical applications in liquid phase microfluidics systems. The same underlying physical phenomena are also encountered in gases and hold great potential for novel particle separation and sorting techniques aimed at industrial and scientific applications. However, considering the physical properties of gases, optimizing the performance of flow-through separators unavoidably requires an understanding of the re-mixing effect of turbulence. In the work presented here we have investigated the effect of turbulence intensity on the separation efficiency of a variable frequency acoustic particle separator featuring a rectangular cross-section with adjustable height. This allows the creation of a standing wave with a variable frequency and number of nodes. The air flow is seeded with alumina particles, 300 nm nominal diameter, and the excitation source is an electrostatic transducer operated in the 50-100 kHz range. In addition to flow and acoustic parameters, the separation efficiency is investigation as a function of geometric parameters such as the parallelism of the resonator walls and the matching between the channel height and the excitation frequency. The measurements made using laser doppler anemometry and light scattering provide guidance for the design of separator configurations capable of advanced separation and sorting tasks with submicron particles.

9:50

2aPA7. Ultrasonic measurements of clays and silts suspended in water. Wayne O. Carpenter (National Center for Physical Acoustics, University of Mississippi, 1 Coliseum Drive, University, MS 38677, wocarpen@olemiss.edu), Daniel G. Wren, Roger A. Kuhnle (Agriculture Research Service - National Sedimentation Laboratory, U.S. Department of Agriculture, Oxford, MS), James P. Chambers, and Bradley T. Goodwiller (National Center for Physical Acoustics, University of Mississippi, University, MS)

Ongoing work at the National Center for Physical Acoustics is aimed at using acoustics to provide monitoring for fine sediments suspended in water. The ultimate goal of the work is to field an acoustic instrument that can monitor fine particle concentration in rivers and streams. Such an instrument would have several advantages over currently available technologies. Expanding upon work from Carpenter et. al (2009), two immersion transducers were placed at a fixed distance to measure attenuation and backscatter from acoustic signals at 10 MHz and 20 MHz propagated through clays (bentonite, illite, and kaolinite) and silt. The resulting data set encompasses a wide range of concentrations (0.01 - 14 g/L) and particle sizes (0.1 - 64 micron diameter particles). Backscatter and attenuation curves for each material across the range of concentrations will be shown and compared to the theoretical attenuation curves developed by Urick (1948). This work has produced a data set for model development using a combination of backscatter and attenuation to allow for single-frequency discrimination between clay and silt particles suspended in water.

10:05–10:20 Break

10:20

2aPA8. Acoustic wave propagation in a channel bifurcated by an elastic partition. Katherine Aho and Charles Thompson (University of Massachusetts Lowell, 1 University Ave, Lowell, MA 01854, katherine_aho@student.uml.edu)

Linear wave propagation in a narrow channel that is axially partitioned by a flexible membrane is examined. The enclosed fluid is excited by the time harmonic displacement in the channel cross-section. The axial variation in the acoustic impedance of the partition gives rise to the generation of evanescent modes in the channel. The effect of these evanescent modes on the vibration of the membrane is of particular interest. It is shown that in the limit of high channel aspect ratio one can model these modes by an effective source distribution along the surface of the membrane. The asymptotic analysis of the source distribution is presented. (NSF Grant 0841392)

10:35

2aPA9. Selected theoretical and numerical aspects of fast volume and surface integral equation solvers for simulation of elasto-acoustic waves in complex inhomogeneous media. Elizabeth Bleszynski, Marek Bleszynski, and Thomas Jaroszewicz (Monopole Research, 739 Calle Sequoia, Thousand Oaks, CA 91360, marek@monopoleresearch.com)

Comparative analysis is considered of two fast FFT-matrix-compression based elasto-acoustic integral equation solvers, employing volumetric and surface formulations, and designed to analyze sound propagation inside a human head; in particular to examine mechanisms of energy transfer to the inner ear through airborne as well as non-airborn path-ways, and to assess effectiveness of noise-protection devices. Verification tests involving the fast surface and volume integral equation solvers are carried out comparing their predictions with those following from an analytical solution of field distribution in an elasto-acoustic layered sphere.Resultsiare presented of representative numerical simulations of acoustic energy transfer to the cochlea for a human head model containing a detailed geometry representation of the outer, middle, and inner ear. The geometry model consists of: (1) the outer surface of the skin surrounding the skull and containing (2) the outer ear represented by its exterior surface, the surface of the auditory canal, and the tympanic membrane modeled as a finite-thickness surface, (3) the middle ear consisting of the system of ossicles and supporting structures, (4) the skull described by its external surfaces and including (5) a set of surfaces representing the inner ear (boundaries of the cochlea, the vestibule, and the semi-circular canals). *This work is supported by a grant from US Air Force Office of Scientific Research.

10:50

2aPA10. A modeling and simulation suite for design of buried object scanning sonars. Hunki Lee, Eunghwy Noh (Mechanical Engineering, Yonsei University, Seoul, Republic of Korea), Kyounghun Been, Hongmin Ahn, Wonkyu Moon (Mechanical Engineering, Pohang University of Science and Technology, Pohang, Republic of Korea), and Won-Suk Ohm (Mechanical Engineering, Yonsei University, 50 Yonsei-ro, Seodaemun-gu, Seoul, Seoul 120-749, Republic of Korea, ohm@yonsei.ac.kr)

In this talk we highlight a work in progress, concerning the development of a comprehensive modeling and simulation (M&S) suite for design of buried object scanning sonars. The M&S suite is expected to cover almost all aspects of physical and engineering acoustics involved in the design process, ranging from transducers, sound propagation, sediment acoustics, backscattering by buried objects, to sonar image processing. The overview of the M&S suite is given along with a preliminary demonstration in the context of a cylindrical object buried in sandy sediment. [This work was conducted in the Unmanned Technology Research Center (UTRC) sponsored by the Defense Acquisition Program Administration (DAPA) and the Agency for Defense Development (ADD) in the Republic of Korea.]

11:05

2aPA11. Multi-frequency modes in dispersive media. Craig N. Dolder and Preston S. Wilson (Department of Mechanical Engineering & Applied Research Laboratories, University of Texas, 10000 Burnet Road, Austin, TX 78758, dolder@utexas.edu)

A common phenomenon in acoustics is the existence of multiple eigenfunctions (mode shapes) corresponding to the same eigenvalue (frequency), which is known as degeneracy. In highly dispersive media the opposite can occur, whereby a single eigenfunction corresponds to multiple eigenvalues. Several ways to visualize the source of, and interpret the physical meaning of, this phenomenon are presented. Instances of this phenomenon occurring in analytical models and experiments are used as examples.

11:20

2aPA12. Estimating the acoustic impedance of the ground using signals recorded by a 3D microphone array. W. C. Kirkpatrick Alberts (RDRL-SES-P, US Army Reserach Laboratory, 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net)

In applications where there is a need to accurately classify impulsive acoustic events, the impedance of the ground can significantly alter the reflected wave such that the superimposed direct and reflected waves could lead to erroneous classifications. If the phase and amplitude changes attributable to impedance ground are considered, knowledge of the ground impedance near every array site can mitigate classification errors caused by impedance ground. Under controlled experimental conditions, the ground impedance in the vicinity of an array can be deduced using standard methods. However, when an array is fielded in an uncontrolled environment, alternative ground impedance estimation techniques must be explored. Soh et al. [J. Acoust. Soc. Am, 128(5), 2010] demonstrated that the impedance of the ground could be directly determined using a pair of vertically spaced microphones and an impulsive source 400 m from the microphones. With some increase in complexity, this direct method of determining ground impedance can be applied to recordings from typical three dimensional acoustic arrays. Estimates of the acoustic ground impedance obtained directly from recordings by microphones distributed in a 1 m radius tetrahedral array will be discussed.

11:35

2aPA13. Evaluating duplex microstructures in polycrystalline steel with diffuse ultrasonic backscatter. Hualong Du and Joseph A. Turner (University of Nebraska-Lincoln, Nebraska Hall, Lincoln, NE 68588, hualong.du@huskers.unl.edu)

The performance of metallic components is governed in large part by the microstructure of the base material from which the component is manufactured. In this presentation, diffuse ultrasonic backscatter techniques are discussed with respect to their use for monitoring the microstructure of polycrystalline steel as a result of the manufacturing process. To improve the mechanical properties, the surface of polycrystalline steel is quenched, a process which transforms the initial phase to a pearlite phase within grains. A diffuse ultrasonic backscatter model is developed that includes the duplex microstructure through the addition of an additional length scale in the two-point spatial correlation function. This function defines the probability that two randomly chosen points will fall into the same grain and/or same crystallite. The model clearly shows the dependence of the diffuse ultrasonic backscatter signal with respect to frequency, average grain size and lamellar spacing of the crystallites. Experimental results are used to show how the two length scales can be extracted from the measurements. The spatial variation of the microstructure with respect to depth from the quench surface is also examined. These diffuse ultrasonic techniques are shown to have the sensitivity to deduce the duplex microstructure throughout the sample.

11:50

2aPA14. Elastic properties of coarse grained lead-free solder alloys. Josh R. Gladden and Sumudu Tennakoon (Physics & NCPA, University of Mississippi, University, MS 38677, jgladden@olemiss.edu)

Because of health and environmental concerns about lead, lead-free solder alloys in most consumer electronics have been required in the European Union since 2006. Many of these alloys are prone to mechanical failure over time, leading to less reliable circuitry. The source of these failures is not well known and many have conjectured that the coarse grained alloys become more brittle over time when exposed to elevated temperatures (~100 °C). Our group, in collaboration with Cisco Systems, has recently studied the effects of aging on the mechanical properties of Sn-Ag-Cu (SAC) solder alloys using using both resonant ultrasound spectroscopy (RUS) and conventional pulse-echo methods. With grain sizes on the order of 100's of microns, the heterogeneity of these alloys present a particular problem for RUS and interpretation of pulse-echo data. Resonance data exhibiting the effect of the heterogeneity will be presented and discussed. Elastic moduli derived from pulse-echo methods as a function of temperature and isothermal aging time will also be shown.

Session 2aSAa

Structural Acoustics and Vibration: Session in Honor of Preston W. Smith, Jr.

Allan D. Pierce, Cochair
P.O. Box 339, East Sandwich, MA 02537

J. Gregory McDaniel, Cochair
Mechanical Engineering, Boston Univ., Boston, MA 02215

Chair's Introduction—8:30

Invited Papers

8:35

2aSAa1. Preston Smith and waves in a cylindrical shell. James G. McDaniel (Mechanical Engineering, Boston University, 110 Cummington Street, Boston, MA 02215, jgm@bu.edu)

In 1955, Preston Smith wrote a landmark paper on free waves in a cylindrical shell. That paper described the displacement components and dispersions of the flexural, shear, and longitudinal waves that propagate in helical directions in the wall of the cylinder. The present author first met Preston over forty years later, when the hand calculations of 1955 had been transferred to a computer. After that meeting in 1996, the group at BBN was interested in how these waves reflect from terminations. Preston developed an approach for solving this problem and worked with the group to implement it using finite element analysis of a cylindrical shell with terminations. The essence of the approach was to mechanically excite different mixtures of waves using different excitations. The amplitudes of incident waves were related to the amplitudes of reflected waves by a reflection matrix. This matrix quantified the wave conversion that occurs at shell terminations. In addition, Preston formulated the reciprocity conditions that must be satisfied by the reflection matrix. His work revealed the most important physics of a very complex system. The present lecture will describe his approach and will highlight his profound style of analysis.

8:55

2aSAa2. Vibrational response features of a locally excited fluid-loaded plate with attached mass-spring oscillator systems. David Feit (Treasurer, Acoustical Society of America, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502, feit.d@att.net)

Preston Smith and others have examined the radiation features of locally excited plates that are periodically supported by inertial masses. This presentation looks at the vibrational response and on-surface pressure field of a locally excited fluid-loaded plate that has one or more attached mass-spring oscillators. The analysis makes use of the "rational function approximation" (RFA) representation of the fluid loading effect first introduced by Diperna and Feit (J. Acoust. Soc. Am. **114**(1), July 2003, pp. 194-199).

9:15

2aSAa3. Wave number filtering on a finite periodically supported plate: Implications on the vibration field, radiated power, and validity of SEA. Robert Haberman (Raytheon IDS, 11 Main Street, Mystic, CT 06355, Robert_C_Haberman@Raytheon.com)

Preston Smith published and presented a number of papers on wave propagation and sound radiation from periodically supported plates. An excellent reference is, "Radiation from Periodically Supported Fluid-Loaded Plates", BBN Report No. 3999, January 1979. In these papers he identifies the physics of Bloch wave radiation, coherent scattering from ribs and spatial attenuation. As an extension to Preston's work, the problem of propagation of a local isotropic wave field on a finite periodically supported plate is considered. The specific question to be addressed is: do the rib supports provide wave number filtering of the isotropic field as it propagates throughout the plate-stiffener system? This question is answered via a series of analytical and finite element models. The implications on radiated power and validity of the SEA isotropic vibration field assumption will be discussed.

9:35

2aSAa4. A review of recent advances in vibro-acoustic system response variance determination in statistical energy analysis: A tribute to Preston Smith, Jr. Robert M. Koch (Chief Technology Office, Naval Undersea Warfare Center, Code 1176 Howell Street, Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil)

Since the pioneering work of Preston Smith, Jr. and Richard Lyon in 1959 in the development of the theory of Statistical Energy Analysis (SEA), followed by many others in the 1960's on through today, the US Navy has utilized the important SEA vibro-acoustic simulation approach for high frequency self- and radiated-noise predictions of a multitude of undersea vehicles and systems. As a tribute to Preston Smith, this talk will review the current state of research in the determination of the variance/probability distribution about the mean response of a system modeled in SEA. While the subject of system response variance (or confidence interval) has obviously been of interest since the inception of this energy-based statistical method, there has been significant recent research in the literature advancing this area that is worth reviewing. As an additional acknowledgement of Preston Smith's later important work in the area of underwater cylindrical shell acoustics, the current presentation will also revisit the canonical structural acoustic problem of a point-excited, finite cylindrical shell with fluid loading and compare SEA-derived radiated noise level predictions with a variety of different classical analytical and modern day numerical approach solutions (e.g., FEA, EFEA/EBEA, closed form plate and shell theory solutions).

9:55

2aSAa5. Macro ocean acoustics. Henry Cox (Information Systems and Global Solutions, Lockheed Martin, 4350 N. Fairfax Dr., Suite 470, Arlington, VA 22203, harry.cox@lmco.com)

Today, physical insight based on fundamental principles and invariants frequently gives way to precise computations based on sophisticated models with less than perfect inputs. In the spirit of Preston W. Smith it is appropriate to revisit what can inferred from the sound speed profile and simple power flux reasoning without extensive computations. For example, based on the sound speed profile, the angle at the axis, the turning depths, the grazing angles at the surface and bottom, ray angle diagrams, the cycle length, cycle time, the group velocity, the adiabatic invariant and the so-called shallow water invariant all can be parameterized in terms of the phase velocity of each mode or ray. Applications of Preston's ideas to ambient noise near the bottom on the deep ocean, coherence and interference patterns will be discussed.

10:15

2aSAa6. Beam broadening for planar transmit arrays with maximal transmit power constraint. Evan F. Berkman (Applied Physical Sciences Inc., 49 Waltham St, Lexington, MA 02421, fberkman@aphysci.com)

Broad active sonar transmit beamwidth enables wide sector search. Many active search sonars attain wide transmit beamwidth by virtue of a cylindrical or spherical geometry which provides naturally wide beamwidth when all elements are driven in phase. However, planar array geometry is desired for some applications. Arrays often utilize many elements with resulting aperture large compared to an acoustic wavelength in order to provide high power output. However, planar arrays with aperture large compared to an acoustic wavelength will have naturally narrow beams which cannot be appreciably broadened by conventional amplitude shading without sacrifice of total power output and inefficient use of transducer and power amplifier channels. Maintenance of broad sector coverage over large fractional bandwidth with all element channels fully driven for maximum power output over the entire frequency band is also challenging. Thus, in order to maintain efficiency of element usage a scheme has been developed for obtaining a specified broad transmit array beamwidth invariant over a wide frequency range by frequency dependent phasing of array elements with all elements constrained to uniform amplitude gain. The intuitive physical basis for the phasing scheme is described as well as the mathematical results. A numerical example is provided.

10:35

2aSAa7. Statistical characterization of multipath sound channels. Peter Cable (Applied Physical Sciences Corp., 135 Four Mile River Road, Old Lyme, CT 06371, petercable@att.net)

Averaged transmission characteristics for underwater sound channels, using energy flux descriptions, were independently introduced and developed by Leonid Brekhovskikh [Sov. Phys.-Acoust. 11, 126-134 (1965)], David Weston [J. Sound Vib. 18, 271-287(1971)], and Preston Smith [J. Acoust. Soc. Am. 50, 332-336 (1971)]. While most attention in these studies was focused on elucidating averaged transmission loss in range dependent environments, Preston, in particular, also examined the application of energy flux techniques to other statistical characterizations of multipath environments, including channel impulse response and spatial coherence. Preston's incisive physical insight resulted in channel statistical characterizations that were simple to apply, but notably effective for signal processing and sonar studies. My purpose in this talk will be to trace the development of statistical characterizations of multipath channels based on energy flux notions, and to sketch some specific applications of these ideas, such as to source or receiver motion induced acoustic fluctuations.

10:55

2aSAa8. A range recursive algorithm for random sum acoustic propagation loss prediction. Cathy Ann Clark (Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell Street, B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

The work of P. W. Smith, Jr. includes prediction of the averaged impulse response for sound transmission in a shallow-water channel. His predictions are most applicable in situations for which cycle mixing with range results in randomization of phase interference between modes. In this talk, a calculation of random summation propagation loss for these types of conditions is presented. An integral expression approximating the normal mode sum is reformulated through a change of variable to an integral with respect to cycle number. The resultant formulation leads to a recursive relation in the range variable which enables calculations to be simplified significantly. The integral formulation is shown to successfully reproduce propagation loss with range by comparison to measurements in a number of environments.

11:15

2aSAa9. Preston Smith's theory of the effect of heat radiation on sound propagation in gases. Allan D. Pierce (P.O. Box 339, East Sandwich, MA 02537, adp@bu.edu)

The 1996 Trent-Crede Medal encomium by Barger states that Smith's favorite paper was "Effect of heat radiation on sound propagation in gases" (JASA, 1957). Smith stated this also in a private communication some time earlier to the present writer. The paper was prompted by works by Stokes and Rayleigh in which heat effects were modeled by Newton's law of cooling, which presumes that the heat radiation out from a limited region of heated matter is proportional to the difference of the local temperature and that of the surrounding medium. Stokes in 1851 constructed a theory for how this assumption leads to a prediction of the dependence of phase velocity and attenuation on frequency, and this theory was used by Rayleigh in the Theory of Sound to analyze whether acoustical fluctuations were more nearly isothermal or adiabatic. However, as Smith pointed out, apparently for the first time, neither Stokes and Rayleigh fully understood the relevant physics. When the atomic nature of heat radiation within gases is taken into account, the effect of heat radiation (in contrast to the effect of thermal conduction) is negligible at all frequencies. For all frequencies for which the attenuation is small, the acoustic fluctuations are adiabatic.

11:35

2aSAa10. Preston Smith and NASA Contractor Report CR-160. Richard H. Lyon (Consulting, RH Lyon, 60 Prentiss Lane, Belmont, MA 02478, rhlyon@lyoncorp.com)

When I came from a post-doc in England to BBN in 1960, Preston had already been at BBN for a few years. One of his interests was the interaction of structural resonances and reverberant sound fields. He had found that if the damping of the structure were to vanish, the response would not diverge, but reach a limit proportional to the sound pressure alone, independent of structural parameters. It turned out that this limit corresponded to modal energy equality between the sound field and the structure and was consistent with work I had done at Manchester on the energy flow between resonators. The combination of the two approaches was the beginning of SEA, presented to the community in "Sound and Structural Vibration, NASA Contractor Report CR-160 by Preston W. Smith Jr. and Richard H. Lyon", March 1965, the first publication on Statistical Energy Analysis (SEA). Interestingly, the words "Statistical Energy Analysis" did not appear in the report, but the ideas and viewpoint were there.

TUESDAY MORNING, 23 OCTOBER 2012 LIDO, 8:30 A.M. TO 11:45 A.M.

Session 2aSAb

Structural Acoustics and Vibration: Guided Waves for Nondestructive Evaluation and Structural Health Monitoring I

Tribikram Kundu, Cochair
Civil Engineering & Engineering Mechanics, University of Arizona, Tucson, AZ 85721

Wolfgang Grill, Cochair
Institute of Experimental Physics II, University of Leipzig, Leipzig 04312, Germany

Chair's Introduction—8:30

Invited Papers

8:35

2aSAb1. Monitoring of corrosion in pipelines using guided waves and permanently installed transducers. Michael J. Lowe, Peter Cawley, and Andrea Galvagni (Mechanical Engineering, Imperial College London, South Kensington, London SW7 2AZ, United Kingdom, m.lowe@imperial.ac.uk)

Guided Wave Testing (GWT) of pipelines for the detection of corrosion has been developed over about 20 years and is now a well established method worldwide, used mostly in the oil and gas industry. The established approach is as a screening tool: GWT is used to detect the presence of significant reflectors which are then examined locally in detail using conventional methods of NDE. To date most of the equipment has been developed for deployment solely at the time of test. However recent developments include permanently-attached transducers which can be left in place after testing, for example to allow easier access for future testing at difficult locations such as buried pipes. This is enabling a new approach, in which improved sensitivity may be achieved by detecting changes with respect to earlier reference signals, and also continuous monitoring which may follow degradation during service. The presentation will include a summary of the GWT method and discussion of current research for monitoring.

9:00

2aSAb2. Estimation of adhesive bond strength in laminated safety glass using guided mechanical waves. Henrique Reis (Industrial and Enterprise Systems Engineering, University of Illinois at Urbana-Champaign, 117 Transportation Building, 104 South Mathews, Urbana, IL 61801, h-reis@illinois.edu)

Laminated safety glass samples with different levels of adhesive bond strength were manufactured and tested using mechanical guided waves. The adhesive bond strength of the test samples was then also evaluated using the commonly used destructive testing method, i.e., the pummel test method. The interfaces between the plastic interlayer and the two adjacent glass plates are assumed to be imperfect and are modeled using a bed of longitudinal and shear springs. The spring constants were estimated using fracture mechanics concepts in conjunction with surface analysis of the plastic interlayer and of the two adjacent glass plates using atomic force microscopy and profilometer measurements. In addition to mode shape analysis, the phase and energy velocities were calculated and discussed. The guided wave theoretical predictions of adhesion levels using energy velocities were validated using the experimental pummel test results. From the attenuation dispersion curves, it was also observed that the S1 mode exhibits attenuation peaks in specific frequency ranges, and that the attenuation of these peaks is sensitive to the interface adhesion levels. Results show that this guided wave approach is useful in the nondestructive assessment of adhesive bond strength in laminated safety glass.

9:25

2aSAb3. Incorporating expected sparsity of damage into ultrasonic guided wave imaging algorithms. Jennifer E. Michaels and Ross M. Levine (School of Electrical and Computer Engineering, Georgia Institute of Technology, 777 Atlantic Drive, NW, Atlanta, GA 30332-0250, jennifer.michaels@ece.gatech.edu)

Many imaging methods employing ultrasonic guided waves are based upon delay-and-sum algorithms whereby echoes scattered from sites of damage are constructively reinforced after signal addition. Resolution of the resulting images depends upon such factors as the underlying array geometry, spectral content, knowledge of the propagation environment, and incorporation of phase information. For plate-like structures of engineering interest, geometrical features such as edges, cut-outs and fastener holes contribute to signal complexity and can cause significant image artifacts, which hinders detection and localization of actual damage. However, it is reasonable to make the *a priori* assumption that damage is spatially sparse. If this assumption is properly incorporated into imaging algorithms, then the resulting images should also be sparse and thus be easier to interpret. Several algorithms are developed and implemented that are based upon sparse reconstruction methods, and their performance on both numerical and experimental data is evaluated in terms of image quality and computational efficiency.

9:50

2aSAb4. Modeling of nonlinear guided waves and applications to structural health monitoring. Claudio Nucera and Francesco Lanza di Scalea (University of California San Diego, 9500 Gilman Drive, La Jolla, CA 92093-0085, flanza@ucsd.edu)

Research efforts on nonlinear guided wave propagation have increased dramatically in the last few decades because of the large sensitivity of nonlinear waves to structural condition (defects, quasi-static loads, instability conditions, etc...). However, the mathematical framework governing the nonlinear guided wave phenomena becomes extremely challenging in the case of waveguides that are complex in either materials (damping, anisotropy, heterogeneous, etc...) or geometry (multilayers, geometric periodicity, etc…). The present work develops predictions of nonlinear second-harmonic generation in complex waveguides by extending the classical Semi-Analytical Finite Element formulation to the nonlinear regime, and implementing it into a highly flexible, yet very powerful, commercial Finite Element code. Results are presented for the following cases: a railroad track, a viscoelastic plate, a composite quasi-isotropic laminate, and a reinforced concrete slab. In these cases, favorable combinations of primary wave modes and resonant double-harmonic nonlinear wave modes are identified. Knowledge of such combinations is important to the implementation of structural monitoring systems for these structures based on higher-harmonic wave generation. The presentation will also present a specific application of nonlinear guided waves for the monitoring of thermal stresses in rail tracks to prevent buckling.

10:15–10:30 Break

10:30

2aSAb5. Imaging-based quantitative characterization of fatigue crack for structural integrity monitoring using nonlinear acousto-ultrasonics and active sensor networks. Zhongqing Su (The Department of Mechanical Engineering, The Hong Kong Polytechnic University, Office: FG 642, Kinmay W. Tang Building, Hong Kong, MMSU@polyu.edu.hk), Chao Zhou, Li Cheng, and Ming Hong (The Department of Mechanical Engineering, The Hong Kong Polytechnic University, Hong Kong, Kowloon, Hong Kong)

The majority of today's damage detection techniques rely substantially on linear macroscopic changes in either global vibration signatures or local wave scattering phenomena. However, damage in real-world structures often initiates from fatigue cracks at microscopic levels, presenting highly nonlinear characteristics which may not be well evidenced in these linear macroscopic changes. It is of great significance but also a great challenge to quantitatively characterize micro-fatigue cracks without terminating the normal operation of an engineering structure. This is a critical step towards automatic and online structural integrity monitoring (SIM). By exploring the nonlinearities of higher-order acousto-ultrasonic (AU) waves upon interaction with fatigue cracks, a damage characterization approach, in conjunction with use of an active piezoelectric sensor network, was established, with the particular capacity of evaluating multiple fatigue cracks at a quantitative level (including the co-presence of multiple cracks, and their individual locations and severities). Identification results were presented in pixelated images using an imaging algorithm, enabling visualization of fatigue cracks and depiction of overall structural integrity in a quantitative, rapid and automatic manner. The effectiveness of the proposed technique was demonstrated by experimentally characterizing multiple fatigue cracks near rivet holes in aluminium plates.

10:55

2aSAb6. Ultrasonic waves for the inspection of underwater waveguide structures. Elisabetta Pistone and Piervincenzo Rizzo (Civil and Environmental Engineering, University of Pittsburgh, 3700 O'Hara Street, Pittsburgh, PA 15261, pir3@pitt.edu)

The non destructive inspection of immersed structures is popular as it minimizes unexpected and costly failures of important marine structures. In this paper we present a non-contact laser/immersion transducer technique for the inspection of underwater waveguide structures. The technique uses laser pulses to generate leaky guided waves and conventional immersion transducers to detect these waves. To prove the feasibility of the proposed methodology, a laser operating at 532 nm is used to excite leaky guided waves in a plate subjected to different damage scenarios. The plate is immersed in water at constant temperature and damage is first simulated using different weights located in the region of interest, i.e. between the point of the laser illumination and the immersion transducers. Damage is also simulated by engraving a series of notches on the face of the plate exposed to the probing system. The waveforms are then processed using the joint time-frequency analysis of the Gabor wavelet transform, statistical features and advanced signal processing techniques to identify and locate the presence of the defects. The findings show that the probing system and the signal processing algorithm used are able to detect differences between pristine and damaged conditions.

11:20

2aSAb7. Defect visualization in pipes using a longitudinal guided wave mode. Hyeonseok Lee (Korea Advanced Institute of Science and Technology, Daejeon, Republic of Korea), Hyun Woo Park (Department of Civil Engineering, Dong-A University, Busan, Republic of Korea), and Hoon Sohn (Department of Civil and Environmental Engineering, Korea Advanced Institute of Science and Technology, 291 Daehak-Ro, Yuseong-Gu, Daejeon 305-701, Republic of Korea, hoonsohn@kaist.ac.kr)

Recently, defect visualization techniques based on guided waves have been developed for pipe inspection. This study advances existing defect visualization techniques in two ways: (1) a fiber-guided laser ultrasonic system, which can operate under high radiation and temperature environments, is used to generate and measure broadband guided waves, and (2) a longitudinal mode instead of a torsional mode is used to detect axial cracks and wall thinning. Using optical fiber probes installed along a circumferential direction of a pipe with equal spacing, a pure longitudinal mode, L(0,2), is launched by axisymmetrically exciting a pipeline structure. The generated L(0,2) subsequently interacts with scattering sources such as defects or pipe boundaries and generate reflected L(0,2) and higher-order modes, L(n,2), (n>0). The reflected modes, L(0,2) and L(n,2), are measured in a pulse-echo manner using the same fiber probes and synthetically processed. By back propagating the dispersive L(0,2) and L(n,2) modes in time and space, this study reconstructs dispersion-compensated L(0,2) and L(n,2) at the scattering sources and thereby visually locates the defects. Numerical simulation and experimental studies are performed to validate the effectiveness of the proposed technique.

TUESDAY MORNING, 23 OCTOBER 2012 TRUMAN A/B, 8:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Cross-Language Production and Perception of Speech (Poster Session)

Wendy Herd, Chair
Mississippi State University, Mississippi State, MS 39762

Contributed Papers

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

2aSC1. Comparing L1 and L2 phoneme trajectories in a feature space of sound and midsagittal ultrasound tongue images. Keita Sano, Yuichi Yaguchi, and Ian Wilson (University of Aizu, Ikkityo Kamega Fujiwara 198, Aizuwakamatsu, Fukushima 965-0005, Japan, ksano065@gmail.com)

To support the development of pronunciation training systems for non-native (L2) speakers, past research has proposed visualization of a speaker's tongue using ultrasound as feedback showing differences between L2 and native (L1) speakers. However, there has been little or no quantitative assessment combining temporal variation of speech sounds and ultrasound tongue images. We propose a mining method to analyze such temporal differences between L1 and L2 speakers. We firstly construct two eigenspaces: one made from feature vectors of speech sounds using Spectrum Vector Field (SVF) and the other from ultrasound tongue images using Histogram of Oriented Gradients (HOG). Next, we compare the movements of L1 and L2 trajectories. Furthermore, we model the connection of phonemes by finding tongue shapes from adjacent speech sounds, and we indicate the differences between L1 and L2 speakers to make a clear intermediate representation from the feature space. In our experiment, we analyze the differences between L1 and L2 pronunciation by focusing on the temporal trajectories of the feature space. These trajectory differences between L1 and L2 speakers' speech sounds will be presented. We will also present the feature space of ultrasound tongue images that indicate the intermediate tongue shapes mentioned above.

2aSC2. Delayed feedback disrupts optimal strategies during foreign speech sound learning. Bharath Chandrasekaran, Han-Gyol Yi (Communication Sciences and Disorders, University of Texas, Austin, TX 78712, bchandra@mail.utexas.edu), and W. Todd Maddox (Psychology, University of Texas, Austin, TX)

The Competition between Verbal and Implicit Systems (COVIS) model posits that an explicit hypothesis-testing system competes with an implicit procedural-based system to mediate category learning. During early learning, the hypothesis-testing system is dominant, whereas in later learning, the procedural system dominates. In visual category learning, delayed feedback is known to impair the procedural but not the hypothesis-testing system. We tested the COVIS model in natural auditory category learning. Young adult native English speakers learned to categorize lexical tones in Mandarin syllables. Timing of feedback was either immediate (0 s) or delayed (500 or 1000 ms). Consistent with COVIS, delay feedback affected accuracy only in later learning. Further, modeling analysis revealed that participants were more likely to adopt procedural strategies during later learning, but this transition was disrupted by delayed feedback. These results will be discussed in the context of developing methods to optimize foreign speech sound learning.

2aSC3. Training adult learners of English to hear the sounds of English. Charles S. Watson, James D. Miller, and Gary R. Kidd (Research, Communication Disorders Technology (CDT), Inc., Bloomington, IN 47404, jamdmill@indiana.edu)

Adult students of foreign languages frequently claim that native speakers of that language speak too rapidly. This is likely a result of the students' failure to achieve automaticity in recognition of speech sounds necessary for effortless speech perception. Research on the time course of auditory perceptual learning for both speech and non-speech sounds provides strong evidence that adults can, with appropriate training, achieve perceptual skills approximating those of native speakers, although they only rarely do so. Among the few adults who do achieve near-native conversational skills in an L2, many have had intensive recognition practice and training. The Speech Perception Assessment and Training System for students of English as a Second Language (SPATS-ESL) of CDT, Inc. provides such training. SPATS-ESL trains the identification of the 109 most common English syllable constituents (onsets, nuclei, and codas) and the recognition of meaningful sentences spoken by a variety of native speakers. Based on experience with over 200 ESL learners it has been found that near-native performance in the recognition of discrete English speech sounds and meaningful sentences spoken by many talkers is acquired by most ESL students after 15-30 hours of individualized computer-based training. (Watson and Miller are stockholders in CDT, Inc.)

2aSC4. High variability training increases mismatch negativity responses to L2 contrasts. Wendy Herd (English Dept., Mississippi State University, 100 Howell Hall, PO Box E, Mississippi State, MS 39762, wherd@english.msstate.edu), Robert Fiorentino, and Joan Sereno (Linguistics Dept., University of Kansas, Lawrence, KS)

Previous research established that high variability training improves both perception and production of novel L2 contrasts and that training noncontrastive sounds in subjects' L1 results in increased MMN responses to those sounds. However, it is unclear whether training novel contrasts in an L2 also results in increased amplitude of MMN responses to the contrasts. This study trained 10 American English learners of Spanish, for whom tap and /d/ are noncontrastive, to distinguish the phonemic tap-/d/ contrast in Spanish to determine if training also changed MMN responses to those sounds when presented in an oddball paradigm. First, the amplitude of native Spanish speakers' (N=10) MMN response to deviant tap was significantly more negative than to the standard, establishing this paradigm elicited canonical MMN responses. Second, trainees (N=10) and controls (N=10) did not exhibit significantly different responses to deviant and standard tap at pretest. Crucially, this was not the case at posttest. Trainees, like native Spanish speakers, exhibited a significant MMN response to deviant tap compared to the standard at posttest, but controls did not. The emergence of an MMN response in the trainees indicates it is possible to recategorize L1 contrasts when learning an L2. [Supported by NSF 0843653.]

2aSC5. Perception of speech-in-noise for second language learners and heritage speakers in both first language and second language. Michael Blasingame and Ann R. Bradlow (Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, IL 60208, mblasingame@u.northwestern.edu)

This study asks whether speech recognition by bilingual listeners in each of their two languages follow complementary or supplementary patterns. Previous studies showed that early bilinguals are disproportionately affected by adverse listening conditions in L2 (Mayo et al., 1997; Shi et al, 2010; Bradlow & Alexander, 2007), but did not measure performance in L1. The current study extends these results using bilingual performance under adverse listening conditions in both languages to determine whether reduced use of the dominant language by relatively well-balanced bilinguals affects performance in L1 as well as L2. We examine two groups of English-Spanish bilinguals: Spanish learners (SL) and Spanish heritage speakers (SHS). Although both English dominant, crucial differences between these groups are L1 (SL=English, SHS=Spanish) and L1-L2 balance (SL=large imbalance, SHS=relatively balanced). Both groups were presented with sentences in English and Spanish in which final keywords varied on three factors: speech style (clear versus plain/conversational), contextual predictability (high versus low), and signal-to-noise ratio (easier versus harder). Results show SHS do not pattern like monolinguals in either language, yet average performance across both languages is higher than SL. This result suggests the overall system SHS maintain is "larger" than SL, but may be more susceptible to noise.

2aSC6. Idiosyncrasy and generalization in accent learning. Meghan Sumner and Ed King (Linguistics, Stanford University, 450 Serra Mall, Stanford, CA 94035, sumner@stanford.edu)

People understand speech well, despite pronunciation variation. Perceptual learning, where listeners are trained with acoustic features ambiguous between two phonemes and subsequently shift their perceived phoneme boundary, is one way listeners may compensate for variation (Norris et al 2003). These perceptual shifts, however, seem idiosyncratic to one speaker (Eisner & McQueen 2005, Kraljic & Samuel 2005), rarely generalizing to new speakers. We propose that lack of generalization is due to lack of experience mapping phonemes to specific continua; previous work uses continua like [s]-[f], whose midpoints rarely occur in speech. Ambiguous tokens that are never heard in real speech may be perceived as specific to the speaker used in training, preventing generalization. Use of a continuum occurring in accented speech, such as the mapping of English tenseness onto vowel duration, allows manipulation of the idiosyncrasy of the mapping. We train 13 listeners on an idiosyncratic duration mapping (lax to short duration, tense to ambiguous duration) and 11 on an Italian accent pattern (lax to ambiguous duration, tense to long), and test generalization to a different speaker. Listeners generalize the Italian accent, and generalize away from the idiosyncratic pattern. This suggests listeners generalize likely accents, treating unlikely patterns as idiosyncratic.

2aSC7. Functional significance of the acoustic change complex, mismatch negativity, and P3a for vowel processing in native-English listeners and late learners of English. Brett A. Martin, Valerie Shafer (Speech-Language-Hearing Science, Graduate Center of the City University of New York, 365 Fifth Avenue, New York, NY 10016, bmartin@gc.cuny.edu), Marcin Wroblewski (Communication Sciences & Disorders, University of Iowa), and Lee Jung An (Speech-Language-Hearing Science, Graduate Center of the City University of New York, New York, NY)

The acoustic change complex (ACC), mismatch negativity (MMN), and P3a all provide indices of the neural processing of the types of acoustic changes that underlie speech and language perception. The goal of this study was to compare neural correlates of vowel processing for contrasts that have been shown to be easy to perceive in native-English speakers but more difficult for native-Spanish speakers. Processing of a vowel change from /I/ to /E/ was compared in a group of late learners of English and a group of monolingual English listeners (n = 15 per group). Preliminary analyses suggest differences in processing of the vowel change from /I/ to /E/ across the groups. Monolinguals processed the vowel change more rapidly and more accurately than bilinguals. The obligatory response to vowel onset showed a larger N1 for the bilinguals compared to the monolinguals. In addition, group differences were obtained in mean global field power (MMN and P3a were longer for bilinguals) and topography (current source density showed group differences for ACC P2 component, MMN, and P3a). Therefore, ACC, MMN and P3a all showed the effects of native language experience; however, these effects were not identical for each component.

2aSC8. A moving target? Comparing within-talker variability in vowel production between native and non-native English speakers across two speech styles. Catherine L. Rogers, Amber Gordon, and Melitza Pizarro (Dept. of Communication Sciences and Disorders, University of South Florida, 4202 E. Fowler Ave., Tampa, FL 33620, crogers2@usf.edu)

Non-native English speakers may show greater variability in speech production than native talkers due to differences in their developing representations of second-language speech targets. Few studies have compared within-talker variability in speech production between native and non-native speakers. In the present study, vowels produced by four monolingual English speakers and four later learners of English as a second language (Spanish L1) were compared. Five repetitions of six target syllables ("bead, bid, bayed, bed, bad" and "bod"), produced in conversational and clear speech styles, were analyzed acoustically. Fundamental and formant frequencies were measured at 20, 50 and 80% of vowel duration. Standard deviations computed across the five repetitions of each vowel were compared across speaking styles and talker groups. Preliminary data analyses indicate greater within-talker variability for non-native than native talkers. Non-native talkers' within-talker variability also increased from conversational to clear speech for most measures. For some native talkers, within-talker variability was smaller for vowels with near neighbors in the vowel space than for vowels with more spectrally distant neighbors. This correlation was stronger in clear speech for talkers who showed a significant clear-speech intelligibility benefit in production in a related study. Implications for theories of vowel production will be discussed.

2aSC9. Processing reduced speech across languages and dialects. Natasha L. Warner, Daniel Brenner, Benjamin V. Tucker (Linguistics, University of Arizona, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), Jae-Hyun Sung (Linguistics, University of Arizona, Tucson, AZ), Mirjam Ernestus (Centre for Language Studies, Radboud University Nijmegen, Nijmegen, Gelderland, Netherlands), Miquel Simonet (Spanish and Portuguese, University of Arizona, Tucson, AZ), and Ana Gonzalez (Linguistics, University of Arizona, Tucson, AZ)

Normal, spontaneous speech utilizes many reduced forms. Consonants in spontaneous speech frequently have a different manner or voicing than would be expected in clear speech (e.g. /d/ and /ŋ/ in "you doing" both being realized as glides or /dʒ/ in "just" as a fricative), and near or complete deletions are

also common (e.g. the flap in "a little"). Thus, listeners encounter and must process such pronunciations frequently. When speakers and listeners do not share the same dialect or native language, such reductions may hinder processing more than for native listeners of the same dialect. The current work reports a lexical decision experiment comparing listeners' processing of reduced vs. careful stops (e.g. /g/ in"baggy" pronounced as an approximant or as a stop), by several groups of listeners. Results show that listeners from both Arizona and Alberta, Canada can recognize speech by an Arizona speaker with reduced stops, but they recognize the words more easily when stops are clearly articulated. Speech style of the preceding frame sentence has little effect, suggesting that both groups can process the stops regardless of whether surrounding context leads them to expect reduced stops. Additional data from second-language learners and bilingual listeners is currently being collected.

2aSC10. Evidence of Spanish undershoot in a Mexican-American community. Arika B. Dean (English (Linguistics), North Carolina State University, Raleigh, NC 27603, abdean@ncsu.edu)

Previous phonetic work on Spanish vowels has suggested that undershoot does not occur in the Spanish vowel system. Quilis & Esgueva (1983) suggested that Spanish vowels were static with little to no articulatory variation. This study contributes to a growing body of phonetic research attempting to disprove these claims. The data comes from audio interviews conducted in a predominantly Hispanic community in Pearsall, Texas. Acoustic analysis is conducted on all vowels in the Spanish vowel range (/i/, /e/, /a/, /o/, /u/), and each token is measured from the midpoint of the vowel nucleus, as well as onset and offset. Subjects are Spanish speakers of different age, sex, language competency, and socioeconomic status. In response to Lindblöm's (1963) assertion that stress is more significant than duration in determining undershoot, this study raises that question again and finds that stress is more statistically significant than duration in the Spanish undershoot process, contrary to Lindblöm's findings. These results also contradict Willis (2005), who found that duration influences the F1 value for Spanish vowels. The effects of English substrate influence are considered, and the vowels of monolingual Mexican Spanish speakers are analyzed, providing a control against the speakers in the Pearsall community.

2aSC11. Abstract withdrawn

2aSC12. Phonemic processing in compensatory responses of French and English speakers to formant shifted auditory feedback. Takashi Mitsuya (Psychology, Queen's University, 62 Arch Street, Humphrey Hall, Kingston, ON K7L3N6, Canada, takashi.mitsuya@queensu.ca), Fabienne Samson (Psychology, Queen's University, Kingston, ON, Canada), Lucie Ménard (Linguistics, Université du Québec à Montréal, Montréal, QC, Canada), and Kevin G. Munhall (Psychology & Otolaryngology, Queen's University, Kingston, ON, Canada)

Past studies have shown that speakers modify their vowel formant production when auditory feedback is altered in order to make the feedback more consistent with the intended sound. This behavior was thought to minimize acoustic error overall; however, Mitsuya et al. (2011) showed different magnitudes of compensation for altered F1 across two language groups depending on the direction of perturbation. Their results seem to reflect how the target vowel is represented in relation to other vowels around it. From this observation, they proposed that compensation is to maintain perceptual identity of the produced vowel, requiring some phonological processes for error reduction. Yet, the results might have been specific to the language groups examined, and/or unique to F1 production. To generalize Mitsuya et al.'s hypothesis, the current study examined 1) different language groups and 2) F2 production. We compared compensatory behavior of F2 for /ɛ/ among French speakers (FRN) and English speakers (ENG) with decreased F2 feedback. With this perturbation, the feedback sounded like /œ/, which is phonemic in French but not in English. The preliminary data suggest that FRN compensated in response to smaller perturbations and showed greater maximum compensations than ENG.

2aSC13. Production of English vowels by speakers of Mandarin Chinese with prolonged exposure to English. Keelan Evanini and Becky Huang (Educational Testing Service, Rosedale Rd., Princeton, NJ 08541, kevanini@ets.org)

Previous studies of non-native production of English vowels have demonstrated that a native-like attainment of certain distinctions is not guaranteed for all speakers, despite prolonged exposure to the target (e.g., Munro et al. 1996, Flege et al. 1997). The current study examines the applicability of this finding to a group of non-native speakers from the same L1 background (Mandarin Chinese) who are all long-term residents in the USA (7 years minimum) and adult arrivals (> age 18). These non-native speakers (N=36) and a control group of native speakers (N=22) were recorded reading two sets of materials: the Stella paragraph (Weinberg 2012) and five sentences from Flege et al. (1999). Vowel formant measurements were extracted for all tokens from the following three pairs of vowels: [i] ~ [ɪ], [e] ~ [ɛ], and [a] ~ [ʌ]. Euclidean distances between the z-normalized (F1, F2) mean values for the two vowels in each pair for each speaker show that the non-native speakers produce each of the three pairs significantly less distinctly than the native speakers. This finding corroborates previous similar findings and suggests that a speaker's L1 continues to have a strong influence on vowel production, despite long-term exposure to the target.

2aSC14. Quantifying the consonantal voicing effect: Vowel duration in an Italian–American community. Ylana Beller-Marino and Dianne Bradley (Linguistics, CUNY Graduate Center, New York, NY 10010, ybeller@gc.cuny.edu)

Cross-linguistically, vowel duration preceding voiced consonants is greater than that preceding voiceless consonants, all else equal (Chen 1970, Mack 1982). Notably, this consonantal voicing effect is larger for English, a presumed instance of language-specific phonological enhancement of a basic phonetic process. The current study asks whether bilingual speakers maintain separate durational settings, and compares consonantal voicing effects across languages in two participant groups: foreign-born Italian speakers who acquired English as young adults, and US-born speakers from the same community who had simultaneous childhood exposure to Italian and English. The complete materials set employed familiar words, e.g., English *rib*/*rip*; Italian *cubico*/*cupola*, and sampled systematically over vowel height and consonantal place; data reported are drawn from the high-vowel materials subset only. For targets uttered within language-appropriate carrier phrases, both groups exhibited the consonantal voicing effect in each language; both also exhibited the same interaction with language, producing reliably larger effects in English, suggesting that language-specific settings were attained. But crucially, where foreign-born speakers produced a purely phonetic effect in Italian, US-born speakers suppressed phonological enhancement only partially. These findings, plausibly reflecting a degree of interplay between phonologies, are discussed in terms of the circumstances of language learning.

2aSC15. How tongue posture differences affect reduction in coronals: Differences between Spanish and English. Benjamin Parrell (University of Southern California, University of Southern California, Department of Linguistics, GFS 301, Los Angeles, CA 90089, parrell@usc.edu)

It has been suggested that both flapping of English coronal stops [e.g. Fukaya & Byrd, JIPA, 2005; De Jong, JPhon, 1998] and spirantization of Spanish voiced stops [e.g. Parrell, LabPhon, 2012] result from reductions in duration. If this is indeed the case, why would reducing duration in one language lead to spirantization (Spanish) and in another to flapping (English)? We suggest that these differences are the result of different ways the tongue is used to attain oral closure in the two languages: in Spanish, coronal stops are made with blade of the tongue at the teeth; in English, with the tongue tip placed at the alveolar ridge. Because of this difference, the tongue tip is oriented differently in the two languages: upward in English and downwards in Spanish, leading to differing articulatory and acoustic outcomes as duration is shortened. We examine these postural differences using tongue movement data, which allows for direct and dynamic examination of tongue posture and shaping of coronals in both languages. Differences between the two languages will be modeled using TaDA [Nam et al., JASA, 2004] to test how they may lead to different articulatory and acoustic outcomes as duration is reduced. [Supported by NIH.]

2aSC16. American Chinese learners' acquisition of L2 Chinese affricates /ts/ and /ts^h/. Jiang Liu and Allard Jongman (Linguistics, University of Kansas, 1541 Lilac Lane, Lawrence, KS 66044, liujiang@ku.edu)

Many studies on L2 speech learning focused on testing the L1 transfer hypothesis. In general, L2 phonemes were found to be merged with similar L1 phoneme to different degrees (Flege 1995). Few studies examined

whether non-phonemic phonetic categories such as consonantal clusters in L1 help or block the formation of new phonetic categories in L2. The current study examined the effect of L1 English consonantal clusters [ts] (e.g., the ending of the plural noun 'fruits') and [dz] (e.g., the ending of the plural noun 'foods') on learning L2 Chinese affricates /ts/ and /ts^h/. We studied duration and center of gravity (m1) of L2 Chinese affricates /ts/ and /ts^h/ produced by native Chinese speakers, novice American Chinese learners and advanced learners. In terms of duration, both learner groups showed contrast between L2 /ts/ and /ts^h/, which is similar to native Chinese speakers' production. However, for m1, only the advanced learner group showed contrast between L2 /ts/ and /ts^h/, which is similar to native speakers' production while the novice learner group did not show m1 contrast between the two L2 affricates. The duration result can be accounted for by the existence of durational difference between L1 English [ts] and [dz] whereas the lack of m1 contrast between the two L2 affricates for the novice learner group can be accounted for by the absence of m1 difference between L1 English [ts] and [dz].

2aSC17. The production and perception of English stops in a coda position by Thai speakers. Siriporn Lerdpaisalwong (Dept. of Linguistics, University of Wisconsin-Milwaukee, Milwaukee, WI 53201, siriporn@uwm.edu)

This paper reports results from a pilot study on the production and perception of English stops in a coda position by native speakers of Thai with different length of residency (LOR) in the US. This study explores three important issues in second language (L2) acquisition: typological markedness (Eckman, 1997), the relationship between production and perception of speech sounds (Flege, 1988 and 1999), and the length of learning L2 sounds (Flege 1999). There were 13 Thai-speaker participants whose LORs ranged from 1 year to 23 years. They participated in two tasks: sentence reading in a production task, and sentence listening in a perception task. Preliminary results show that participants produced all English voiced stops less accurately than voiceless stops. However, in the perception task, only /g/ was perceived less accurately than voiceless stops. The speakers perceived /b/ better than /k/ and perceived /d/ better than /p/ and /k/. The more accurate the speakers can perceive the sounds, the better they can produce it. The Thai speakers with a longer LOR perceived and produced English stops in the coda position more accurately than those with a shorter LOR. The results found raise our awareness of to which sounds should be paid special attention and the benefit of enough language input. Also, the study suggests the pattern of English stops acquired by native speakers of Thai in both production and perception processes.

2aSC18. Acoustic correlates of stop consonant voicing in English and Spanish. Olga Dmitrieva (Linguistics, Stanford University, 450 Serra Mall, Stanford, CA 94305, dmitro@stanford.edu), Amanda A. Shultz (Linguistics program, Purdue University, West Lafayette, IN), Fernando Llanos (School of Languages and Cultures, Purdue University, West Lafayette, IN), and Alexander L. Francis (Department of Speech, Language, and Hearing Sciences, Purdue University, West Lafayette, IN)

In English, fundamental frequency at the onset of voicing (onset f0) covaries with the Voice Onset Time (VOT) of initial stops and provides an additional perceptual cue to the phonetic feature of voicing, especially when VOT is ambiguous. However, aerodynamic and physiological explanations of the onset f0/VOT relationship suggest that onset f0 should correlate with voicing only in languages such as English that contrast short lag (voiceless unaspirated) with long lag (voiceless aspirated) consonants, and not in languages such as Spanish that contrast prevoiced with short lag stops. Previous perceptual research supports this prediction: Spanish speakers with little English experience do not incorporate onset f0 in making voicing decisions, suggesting lack of a correlation in their ambient language. In contrast, Spanish speakers with extensive experience with English show an English-like pattern of onset f0 use, suggesting that exposure to the English pattern of covariation has influenced their perceptual weighting of these two cues. The present study compares the distribution and correspondence between VOT and onset f0 in syllable-initial bilabial stops ([b] - [p]) in Spanish and English. Implications for the typology of voicing contrasts and perceptual strategies for sound categorization in non-native language environments are discussed.

2aSC19. Modeling learning of the English voicing contrast by Spanish listeners living in the United States. Fernando Llanos (Spanish & Portuguese, Purdue University, West Lafayette, IN), Alexander L. Francis (Speech, Language & Hearing Sciences, Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu), Olga Dmitrieva (Linguistics, Stanford University, West Lafayette, Indiana), Amanda A. Shultz (Linguistics, Purdue University, West Lafayette, IN), and Rachel Chapman (Speech, Language & Hearing Sciences, Purdue University, West Lafayette, IN)

The importance of cue covariation in phonetic learning is explored through four experiments investigating perception of stop consonant voicing. Spanish and English show different uses of voice onset time (VOT; the time between consonant burst release and vocalic voicing onset) in cuing voicing perception. English contrasts short lag (<20 ms) with long lag (>20 ms) stops, whereas Spanish contrasts prevoicing (<0 ms) with short lag (>0 ms) stops. Secondary cues may also differ. In English, VOT and onset f0 (fundamental frequency at voicing onset) are positively correlated, and onset f0 plays a role in voicing perception. In Spanish these properties may not be as strongly correlated, meaning that onset f0 may be less relevant to voicing perception. As predicted, Spanish listeners tested in Spain showed a 0 ms VOT boundary with little use of onset f0, whereas English listeners tested in the US showed a 20 ms boundary with moderate use of onset f0. Significantly, Spanish listeners tested in the US showed an English-like VOT boundary, but made even greater use of onset f0 than did English listeners. Computational Hebbian modeling suggests a role for differences in each groups' experience with specific patterns of VOT/onset f0 covariation.

2aSC20. Perception of American English final consonants by speakers of New York-Dominican Spanish. Shari S. Berkowitz (Communication Disorders, Mercy College, 555 Broadway, Main Hall, G-14-B, Dobbs Ferry, NY 10522, shariellen@gmail.com)

The English language uses many final consonants and final clusters to convey meaning, especially for morphological endings. The Spanish language employs fewer final consonants than the English language, and Caribbean Spanish speakers treat many final consonants as optional. In this experiment, speakers originating from the Dominican Republic (N = 25) participated in a listening task in which they had to identify final consonants in fast and clear sentences in English (stimulus corpus Ito, K, 2011). A small group of native American English speakers was tested, and performed at ceiling. Spanish-speaking participants' performance on the experimental task varied from 40% to native-like accuracy and was statistically different from the native speakers' performance (Mann Whitney $U = 8$, $p < .003$). In addition, Spanish-speaking pariticpants' performance on the final consonant perception task correlated strongly with performance on a standardized aural/oral language battery known as the Versant Test (Pearson Corp, 2011)($r = .76$, $p < .001$); performance also correlated strongly with age of acquisition. The coarticulation of adjacent speech sounds played a role in which consonants were most difficult to perceive. Future directions, including the current testing of speakers of Puerto Rican Spanish, Kannada and Russian, and implications for intervention, will be discussed.

2aSC21. Featural enhancement of Spanish word-initial stops in clarifications of misheard words. Jessamyn L. Schertz (Linguistics, University of Arizona, Tucson, AZ 85721, jschertz@email.arizona.edu)

In an experiment exploring phonetic featural enhancement in Spanish, native speakers were asked to read words aloud, then repeat them when a supposed automatic speech recognizer "guessed" incorrectly (e.g. subject says "basta," computer displays (in Spanish) "Did you say 'pasta'?", subject repeats "basta"). In a previous experiment with the same paradigm, English speakers exaggerated VOT in the second repetition (longer prevoicing for voiced and longer aspiration for voiceless stops) when the incorrect guess was a minimal pair in voicing with the target word. Spanish speakers also had longer prevoicing durations for voiced stops, but unlike English speakers, showed no change in VOT for voiceless stops; in fact, VOT was shorter in the clarification, though not significantly. The differences in how speakers of the two languages manipulated the stops reflects cross-linguistic differences in the phonetic components of the stop contrast. Additionally, Spanish speakers produced fricatives for some of the word-initial voiced stops.

Although Spanish voiced stops are realized as fricatives in many environments, they are not expected to be lenited following a pause. The results of this study confirm that speakers use inventory-specific featural manipulations to clarify contrasts, and demonstrate unexpected variability in post-pausal voiced stops in this dialect of Spanish.

2aSC22. The effects of first-language sound change on second-language speech production. Mi-Ryoung Kim (Practical English, Soongsil Cyber University, Dept. of Practical English, Soongsil Cyber University, 307 Jongno Biz-well, 34 Ikseon-dong, Seoul, OR 110-340, Republic of Korea, kmrg@mail.kcu.ac)

Recent studies have shown that the stop system of Korean is undergoing a sound change in which a consonantal opposition between lax and aspirated stops is merging in terms of voice onset time (VOT) whereas the contrast between the two stops is being maximized in terms of fundamental frequency (f0). This study investigates how the ongoing sound change of acoustic parameters in L1 Korean influences L2 English stop production. Results showed that, unlike the VOT merger in L1 Korean, it does not occur in L2 speech production. In contrast, similar to onset-f0 interaction in L1 Korean, there is a strong onset-f0 interaction in L2 English: voiced-low f0 and voiceless-high f0. Korean English learners use not only VOT but also f0 in contrasting an underlying [voice] distinction. The results suggest that f0 differences between lax and aspirated stops in L1 Korean are transferred to those between voiced and voiceless counterparts in L2 English. The findings are discussed with respect to cross-language phonetic effects and synchronic sound change.

2aSC23. Perception of place-of-articulation contrasts of English word-final consonants in connected speech by Japanese and Korean second language learners. Kikuyo Ito and JungMoon Hyun (Ph.D. Program in Speech-Language-Hearing Sciences, Graduate Center, CUNY, Grad Center, CUNY, 365 5th Ave., New York, NY 10016, kikuyoito@hotmail.com)

An extension of a previous study examining Japanese listeners' perception of place contrasts of English word-final stops in connected speech (Ito, 2010) was carried out by administering the same experiment to Korean listeners. Stimuli embedded in a carrier sentence and produced in fast casual speech were presented in a three-alternative forced choice identification test, adopting minimal triplets (e.g., sip-sit-sick, bib-bid-big, Kim-kin-king) followed by an adverb starting with /p/, /t/, or /k/. Data for 24 Korean listeners were compared with the previous data for 24 Japanese and 24 American English (AE) listeners. Whereas Japanese listeners had exhibited severe difficulty in perceiving place contrasts of nasal and voiceless stops, Korean listeners were expected to have much less perceptual difficulty on those contrasts because of the different L1 phonological rules of final stops. Results revealed that Koreans' response accuracy was much higher than that of Japanese on voiceless stops (Korean 82%, Japanese 67%, AE 90%) and nasal stops (Korean 96%, Japanese 66%, AE 98%), conforming to the predictions. The contrasting performance between Korean and Japanese listeners on nasal stops was especially remarkable, strongly supporting the notion that Japanese listeners' difficulty in perceiving place contrasts of word-final nasals is due to their L1phonological rules.

2aSC24. Non-native perception and production of Basque sibilant fricatives. Melissa M. Baese-Berk and Arthur G. Samuel (Basque Center on Cognition, Brain and Language, Paseo Mikeletegi, 69, Donostia, Guipuzkoa 20009, Spain, m.baese@bcbl.eu)

Differences in perception and production of non-native contrasts are thought to be driven by the relationship between sound inventories of the native and target languages (Best, McRoberts, and Goodell, 2001). The current study examines non-native perception and production of sibilant fricatives and affricates in Basque. Basque has a 3-way place contrast for sibilant fricatives and affricates (apico-alveolar /s̺/ and /ts̺/, lamino-alveolar /s̻/ and /ts̻/, and post-alveolar /ʃ/ and /tʃ/). In contrast, /s/ and /tʃ/ are the only voiceless sibilants that Spanish has in this region. The results suggest that in the case of Basque sibilant phonemes, similarity to an existing contrast (i.e., fricative-to-affricate contrasts) results in better perception and production. Native Spanish speakers performed better on discrimination and repetition of the /s̺/ - /ts̺/ contrast than the /s̺/ - /s̻/ or /s̺/ - /ʃ/ contrasts. Spanish speakers are able to leverage their ability to discriminate and produce /s/ and /tʃ/ in their native language to perceive and produce a new contrast in Basque, even though the contrast in Spanish differs by two features, rather than just one as in Basque. However, the lack of a contrast between sibilant fricatives prevents them from discriminating or producing the fricative-to-fricative contrasts.

2aSC25. Sibilant production patterns in three generations of Guoyu-Taiwanese bilinguals. Ya-ting Shih (Second and Foreign Language Education, The Ohio State University, Columbus, OH 43212, shih.68@buckeyemail.osu.edu), Jeffrey Kallay, and Jennifer Zhang (Linguistics, The Ohio State University, Columbus, OH)

This study investigates the effects of age and language dominance on sibilant production in a bilingual community. Guoyu (Taiwanese Mandarin) has 3 sibilants: alveolar /s/, retroflex /ʂ/ and alveolo-palatal /ɕ/, while Taiwanese (a Southern Min dialect) only has /s/, which is palatalized before /i/ and /io/. Productions of sibilant initial words were elicited using a word repetition task. Subjects were 30 adults in three age bands from 20-80 years, with the oldest being the most Taiwanese-dominant. The spectral centroid was obtained from the middle 40ms of each sibilant, along with the onset F2 of the following vowel. In the low-vowel /a/ context, the youngest speakers clearly separate /s/ in both languages from the Guoyu /ʂ/ and /ɕ/ along the centroid dimension. The /ʂ/ is then separated from /ɕ/ by F2. However, the oldest speakers show no clear separation of these sounds in terms of centroid, although Guoyu /ɕ/ can still be differentiated by F2. Also, the three generations demonstrated differences in the assimilation patterns of palatal sounds. The younger Guoyu-dominant speakers assimilated Taiwanese palatalized one to Guoyu /ɕ/ in both centroid and F2, while older speakers matched the Guoyu /s/~/ɕ/ distinction to that of the Taiwanese pattern.

2aSC26. Acoustic and perceptual similarities between Effutu and English fricatives: Implications for English as a second language. Charlotte F. Lomotey (Texas A&M University, Commerce, TX 75428, cefolatey@yahoo.com)

Volin & Skarnitzl (2010) describe a foreign accent as a set of pronunciation patterns, at both segmental and suprasegmental levels, which differ from pronunciation patterns found in the speech of native speakers (p.1010). Not only can these pronunciation patterns differ, they can also be similar in many ways. These similarities can be perceptual, acoustic and auditory, especially at the segmental level. This study investigates the acoustic and perceptual similarities between the fricatives /s/, /f/ and /ʃ/ of Effutu, a dialect of Awutu, and their English counterparts in the context of /a/ and /i/. Duration and spectral peak frequency are measured in order to achieve this. A discrimination task, Same-Different task, was administered to investigate listeners' perceived similarity (or difference) judgments between the pairs of fricatives. Preliminary findings show that there are perceptual and acoustic differences in the durations of these segments cross-linguistically. This study contributes to cross-linguistic investigation of fricatives, and to second language acquisition. The findings also show that the use of acoustic and perceptual cues helps to establish differences between speech sounds in different languages, and that, ESL teachers can use these to develop appropriate ways of teaching English sounds to learners.

2aSC27. Perception and production of second language sound inventory by English-speaking learners of Korean. Hanyong Park (Department of Linguistics, University of Wisconsin-Milwaukee, Curtin Hall 523, P.O. Box 413, Milwaukee, WI 53201, park27@uwm.edu)

This study investigates the perception of L2 sound inventory and its comparison with L2 production by eleven adult English-speaking novice learners of Korean in a classroom setting. We examined the perceptual identification and production accuracy of Korean consonants and vowels: eight monophthongs /i e ɛ ɨ u o ʌ ɑ/ both in isolation and following /p t k/ contexts, and fourteen consonants /p p' pʰ t t' tʰ s s' c c' cʰ k k' kʰ/ with /ɑ/ in word-initial position. Overall results indicated that most learners were better at production than perception. Such tendency was more apparent for consonants (except /s/) than vowels, for many learners exhibited a high performance in both perception and production of vowels. Results also showed that learners with more accurate production tended to exhibit more accurate perceptual identification. However, such observation applied only to vowels. Further, learners often had difficulty in both production and perception for the same vowels.

Findings suggest rate differences in L2 sound learning; learning takes longer in perception than in production, and in consonants than in vowels. Findings also suggest that production-perception link is stronger in L2 vowel development, at least in the case of English speaking learners of Korean.

2aSC28. Phonetic accommodation after passive exposure to native and nonnative speech. Midam Kim and Ann R. Bradlow (Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, IL 60201, midamkim@gmail.com)

We investigated native English talkers' phonetic accommodation to a native or nonnative model talker in a passive auditory exposure setting. We performed a phonetic accommodation experiment, following the procedure of Goldinger & Azuma (2004). Specifically, the imitators read monosyllabic words, disyllabic words, and sentences before and after perceptual exposure to the stimuli. We found evidence of phonetic convergence both to native and nonnative model talkers from various acoustic measurements on words and sentences, and dynamic time warping analyses and XAB perception tests on sentences. We also found that dialect mismatch between participants and native model talkers inhibited phonetic convergence in some acoustic measurements. Additionally, the distances between model talkers and participants along the acoustic measurements before auditory exposure positively affected their degrees of phonetic convergence, regardless of the direction of the change; the farther the acoustic distance was before the auditory exposure, the larger the degree of phonetic convergence was. Moreover, the imitators generalized their accommodation patterns from exposed to unexposed items. Finally, XAB perception tests with the sentences revealed that imitators of all model talkers were perceived as converging towards their model talker, and importantly, this pattern of perceived accommodation was predicted by most of the sentence-based acoustic measurements.

2aSC29. Factors affecting the perception of foreign-accented speech by native and non-native listeners. Terrin N. Tamati (Linguistics, Indiana University, Bloomington, IN 47401, ttamati@indiana.edu)

Previous research has shown that several factors influence the perception of foreign-accented speech. Beyond talker-related factors, such as native language, length of residency, and age of acquisition, other factors, such as listener experience, listening context, and lexical characteristics, play an important role. To futher investigate these issues, the current study explored the perception of foreign-accented speech by native speakers of American English and Korean learners of English. In an accent rating task, listeners evaluated English sentences produced by native and non-native speakers (Korean and Mandarin) for strength of accent. Sentences contained three key words that varied by lexical frequency (high or low) and phonological neighborhood density (high or low). The same listeners also completed a sentence recognition task with a similar set of materials in which they listened to sentences and typed in the words they recognized. Results showed that lexical frequency and neighborhood density, overall, significantly influenced perceived accentedness and recognition accuracy for both groups. However, these effects were mediated by the native language of the talker and listener. These findings support previous research showing lexical frequency and density effects in the perception of foreign-accented speech and suggest that these effects may interact with talker and listener background.

2aSC30. Processing interactions between segmental and suprasegmental information in English and Mandarin Chinese. Mengxi Lin (Linguistics, Purdue University, West Lafayette, IN), Alexander L. Francis (Speech, Language & Hearing Sciences, Purdue University, SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu), Fernando Llanos (Spanish & Portuguese, Purdue University, West Lafayette, IN), Olga Dmitrieva (Linguistics, Stanford University, West Lafayette, Indiana), and Rachel Chapman (Speech, Language & Hearing Sciences, Purdue University, West Lafayette, IN)

In this study, a Garner selective attention task is used to identify cross-linguistic differences in attention to vowels, consonants and tones. In previous research, Tong et al. (2008) reported that, in Mandarin Chinese, consonantal and vocalic variability interfered more with tone processing than vice versa (asymmetric integrality), in contrast with the findings of earlier studies (Lee & Nusbaum, 1993; Repp & Lin, 1990). However, while these earlier studies examined both English and Mandarin Chinese listeners, Tong et al. (2008) did not study English speakers because of cross-linguistic differences in tone discrimination. The present study extends this work to examine interactions between these properties in English as well as Mandarin Chinese, using stimuli in which the consonantal, vocalic and tonal differences are linguistically meaningful in both languages, and normalizing for cross-linguistic differences in discriminability. It is predicted that Chinese results will replicate those of Tong, et al. (2008), while English listeners may show more symmetric integrality between segmental and tonal information than in previous studies since these pitch contours are prosodically meaningful and corrected for discriminability. Results will be discussed with respect to the role that linguistic knowledge plays in determining processing dependencies between segmental and suprasegmental information.

2aSC31. Delayed use of fundamental frequency (F0) rise in non-native speech segmentation. Caitlin E. Coughlin and Annie Tremblay (Linguistics, University of Kansas, 541 Lilac Lane Blake Hall, Lawrence, KS 60045-3129, atrembla.illinois@gmail.com)

Research has shown that second/foreign language (L2) speech segmentation is less efficient than native language (L1) segmentation, because L2 learners cannot suppress L1 segmentation routines. This study uses visual-world eye-tracking to determine whether English learners of French can learn to use F0 for locating word-final boundaries in French. F0 rise is often word-initial in English but often word-final in French. Native French speakers and English learners of French heard sentences where lexical competitors were created between the target noun and the following adjective (stimulus: chat grincheux 'cranky cat'; target: chat 'cat'; competitor: chagrin 'sorrow'). The target was either accented or unaccented, and the stimuli were either natural or resynthesized (swapped F0 between accented and unaccented targets). Participants selected the word they heard from given options (target, competitor, distracters), and fixations were recorded from target-word onset. Accuracy (word selection): learners, but not natives, were more accurate for target words with F0 rise than without it. Fixations: natives, but not learners, showed higher differential proportions of target and competitor fixations for target words with F0 rise than without it. Proficiency did not interact with the variables. This suggests that L2 learners' use of F0 rise is delayed compared to that of natives.

2aSC32. Cross-language assimilation of lexical tone. Jennifer Alexander and Yue Wang (Department of Linguistics, Simon Fraser University, Robert C Brown Hall Bldg, 8888 University Drive, Burnaby, BC V5A 1S6, Canada, jennifer_alexander@sfu.ca)

We extend to lexical-tone systems a model of second-language perception, the Perceptual Assimilation Model (PAM) (Best & Tyler, 2007), to examine whether/how native-language lexical-tone inventory composition influences perception of novel tone. Native listeners of Cantonese, Thai, and Mandarin perform a tone mapping-rating assimilation task. Listeners hear CV syllables bearing all tones of Cantonese, Thai, Mandarin, and Yoruba - languages with different tone inventories. They (1) map the tone they hear to the nearest native tone category, and (2) provide a goodness rating on a 5-point scale (5 = perfect). As predicted by the PAM, listeners assimilated non-native tones to the phonetically-closest native tone categories. Listeners attended primarily to pitch-contour, and secondarily to pitch-height, contrasts for the mappings. E.g., Mandarin listeners assimilated the Thai high "level" (phonetically mid-to-high-rising) tone to Mandarin rising tone 76% of the time, and to Mandarin high-level tone only 22% of the time. Also as predicted, all novel tones did not assimilate equally well to native categories; mappings received ratings between 2.9-4.1, averaging 3.5. The groups' different patterns of results indicate that novel-tone perception is influenced by experience with the native-language tone inventory, and that listeners attend to gradient phonetic detail to assimilate novel tones to native-tone categories. This work is supported by NSF grant 0965227 to J.A.

2aSC33. Effects of acoustic and linguistic aspects on Japanese pitch accent processing. Xianghua Wu, Saya Kawase, and Yue Wang (Linguistics, Simon Fraser University, 8888 University Drive, Burnaby, BC V5A1S6, Canada, xianghua_wu@sfu.ca)

This study investigates the hemispheric processing of Japanese pitch accent by native and non-native listeners. The non-natives differ in their first (L1) and second (L2) language experience with prosodic pitch, including Mandarin (tonal L1) and English (non-tonal L1) listeners with or without

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Japanese learning experience. All listeners completed a dichotic listening test in which minimal pairs differing in pitch accent were presented. Overall, the results demonstrate a right hemisphere lateralization across groups, indicating holistic processing of temporal cues as the pitch accent patterns span across disyllabic domain. Moreover, the three pitch accent patterns reveal different degrees of hemispheric dominance, presumably attributable to the acoustic cues to each pattern which involve different hemispheric asymmetries. The results also reveal group difference, reflecting the effects of linguistic experience. Specifically, the English listeners with no Japanese background, compared to the other groups, exhibit greater variance in hemispheric dominance as a function of pitch accent difference, showing a greater reliance on acoustic cues when linguistic information is lacking. Together, the findings suggest an interplay of acoustic and linguistic aspects in the processing of Japanese pitch accent but showing a more prominent acoustic influence. [Research supported by NSERC.]

2aSC34. Tonal adaptation of English loanwords in Mandarin: The role of perception and factors of characters. Li-Ya Mar and Hanyong Park (Department of Linguistics, University of Wisconsin-Milwaukee, 4810 Marathon Dr., Madison, WI 53705, liyamar@uwm.edu)

The present study investigates the role of orthography and perceptual similarity between the English stress and the Mandarin tone during the borrowing process of English words among Taiwanese Mandarin speakers. We had 7 Mandarin speakers transliterate 40 unfamiliar disyllabic US city names using Chinese characters. Based on the results, we created 28 stimuli consisting of possible Chinese borrowings and the target English word for AXB identification tasks. Then, we had the subjects choose the more similar Chinese form to an English target word after the auditory presentation of the stimuli, first without and second with written representations of the stimuli on different days. The transliteration results indicate that a stressed syllable is usually adapted with tones with a high pitch. The AXB task results show a character frequency and a semantics override perceptual similarity; when the adapted forms include characters used infrequently or with negative meanings, another form is chosen despite the fact that it is not the most perceptually similar form. The findings suggest that perceptual similarity mapping takes place first and other factors, such as semantics or the character frequency, come into play when the output contains an infrequently-used or semantically-negative character in tonal adaptation.

2aSC35. Orthography modulates lexical recognition in a second language. Christine E. Shea (Spanish and Portuguese, University of Iowa, 412 Phillips Hall, Iowa City, IA 52242, cesshea@yahoo.com)

We use a cross-modal masked priming paradigm to investigate a) whether orthography is always activated during lexical recognition and b) when activated, whether orthography influences the perception of allophonic variants by adult L2 learners. L1 Spanish and L2 Spanish learners (n=60) were exposed to written Spanish primes with ‘b’ ‘d’ or ‘g’ in intervocalic position. In Spanish, the positional phones corresponding to these orthographic symbols are voiced fricatives [β ð ɣ]; in English they are voiced plosives. In the matched prime trials, written primes were paired to auditory targets with the expected voiced fricative (lado ‘[laðo] ‘side’). For the unmatched prime trials, the auditory target had medial plosives ([lado]. Orthographic prime durations were either 33ms (implicit, Condition 1) or 67ms (explicit, Condition 2). Accuracy and reaction times were registered on lexical decision to the auditory target. Preliminary RT results indicate a three-way interaction among group, trial type (matching or unmatching) and prime condition. Follow-up tests revealed a significant difference for the L2 listeners for prime conditions: significantly longer RTs were registered for the ‘matching’ trials when the orthographic prime was visible (Condition 2). These results suggest that L2 lexical recognition is modulated by orthographic information when it is explicitly available.

TUESDAY MORNING, 23 OCTOBER 2012 — BENNIE MOTEN A/B, 8:15 A.M. TO 10:45 A.M.

Session 2aSP

Signal Processing in Acoustics: Methods for Underwater Acoustic Parameter Estimation and Tracking at Low Signal-to-Noise Ratios

Paul J. Gendron, Chair
Maritime Systems Div., SSC Pacific, San Diego, CA 92152

Chair’s Introduction—8:15

Invited Papers

8:20

2aSP1. Information-based performance measures for model-based estimation. Edmund J. Sullivan (Prometheus Inc., 46 Lawton brook Lane, Portsmouth, RI 02871, ejsul@fastmail.fm)

Classical estimation is conventionally evaluated via the Cramer-Rao Lower Bound on the estimate. When prior information is available, Bayesian estimation can be used if this information is available in statistical form. However, when the prior information is in the form of a physical model, such as in a tracking scheme, it is not clear how much improvement will be provided, since it is not in statistical form. Here it is shown how this problem can be dealt with using the Fisher information matrix by introducing the model into a Kalman estimator. Since the state error covariance provided by a steady-state Kalman estimator is the inverse of the Fisher matrix, it directly provides a statistical measure of the information provided by the model. Then by relating the Fisher information matrix to the Kullback-Liebler distance, it is shown how the Fisher matrix is scaled to provide its information in bits. The model can then be evaluated as to how much information it provides to the estimator. An example using a moving towed array as a bearing estimator will be presented. It will be quantitatively shown that inclusion of the array motion in the estimator will improve the estimation performance

8:40

2aSP2. A physical statistical clutter model for active sonar scenarios with variable signal-to-noise ratios. Roger C. Gauss and Joseph M. Fialkowski (Acoustics Division, Naval Research Laboratory, 4555 Overlook Ave., S.W., Washington, DC 20375-5350, roger.gauss@nrl.navy.mil)

Active sonar classification algorithms need to be robust in preventing operator overload while not being misled by false targets. This talk describes a new 3-parameter statistical sonar clutter model that not only provides a physical context for relating the characteristics of normalized matched-filter echo-data distributions to scatterer attributes, but scatterer information that is largely independent of its peak signal-to-noise ratio (SNR) value. It extends our 2-parameter Poisson-Rayleigh model (Fialkowski and Gauss, IEEE JOE, 2010) by adding a quantitative measure of scatterer spatial dispersion to its measures of scatterer density and relative strength. Maximum likelihood estimates of the clutter model's 3 parameters were derived from mid-frequency (1-5 kHz) shallow-water active sonar data containing returns from biologic, geologic and anthropogenic objects with differing spatial and scattering characteristics. The resulting clutter model's probability density functions not only fit the non-Rayleigh data well while displaying an insensitivity to SNR, but the dispersion parameter values were consistent with the known spatial characteristics of the scatterers and the values' ping-to-ping variance correlated strongly with clutter object class, all of which are encouraging with regard to developing robust physics-based active classification algorithms. [Work supported by ONR.]

9:00

2aSP3. Least squares channel estimation and adaptive equalization at low signal to noise ratios. James C. Preisig (AOPE, WHOI, MS #11, Woods Hole, MA 02540, jpreisig@whoi.edu)

Least squares based adaptive algorithms are among the most commonly used techniques for both channel estimation and adaptive equalization using signals that have propagated through underwater acoustic channels. Such channels are often characterized by long delay spreads meaning that the impulse response of the channel contains many "taps" or "weights" to be estimated or accommodated. In addition, the channel is often time varying which limits the duration of the averaging window that can be used by the algorithm's adaptation process. At low SNRs, these two factors pose a significant challenge to algorithm performance. This challenge is particularly severe in the context of adaptive equalization where the use of multichannel equalizers is often required to achieve reliable performance and the traditional approach of joint optimization of the feedforward filter tap weights on all receiver channels results in large dimensional optimization problems. This talk will contrast and compare the impact of low SNR on least squares based channel estimation and adaptive equalization algorithms. The role of dimensionality reduction will be examined and the response of multichannel equalizers to different signal and noise environments as well under different equalizer configurations will be examined.

9:20

2aSP4. Pulse compression in striation processing—Acoustic invariant as seen in the time domain. Paul Hursky (HLS Research Inc, 3366 North Torrey Pines Court, Suite 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

The acoustic invariant is well known to produce broadband interference or striation patterns in spectrograms. These have been used for a variety of applications, including geo-acoustic inversion and target tracking, in both passive and active settings. The processing to extract parameters from these broadband interference patterns has typically been performed on the spectrograms. However, spectrogram striations have energy that is spread across a wide band. This paper will present a time domain approach, in which the striations are pulse-compressed via a correlation process, before extracting the parameters of interest. Narrowband signals pose an interesting conundrum for striation processing - they are typically much stronger iin level than the underlying broadband interference pattern, and can be mistaken for striations, thus corrupting the parameter extraction. As in time delay estimation, pre-whitening is needed to suppress narrowband components. At the same time, the narrowband components best reveal the underlying interference pattern, albeit in a very narrow band, because they are of such high SNR. We will discuss how to exploit this information content as well.

Contributed Papers

9:40

2aSP5. Exploiting differences in underwater acoustic signal and noise distributions to improve signal detection in low signal-to-noise ratio. Andrew T. Pyzdek, R. L. Culver, and Brett E. Bissinger (Applied Research Laboratory, The Pennsylvania State University, PO Box 30, State College, PA 16804, atp5120@psu.edu)

Traditional models for acoustic signals and noise in underwater detection utilize assumptions about the underlying distributions of these quantities to make algorithms more analytically and computationally tractable. Easily estimated properties of the signal, like the mean amplitude or power, are then calculated and used to form predictions about the presence or absence of these signals. While appropriate for high SNR, quantities like the mean amplitude may not give reliable detection for SNR at or below 0 dB. Fluctuation based processors, utilizing additional statistics of received pressure, offer an alternative form of detection when features of the received signal beyond changes in mean amplitude are appreciably altered by the presence of a signal. An overview of fluctuation based processing will be given, with a focus on the underlying statistical phenomena that grant this method efficacy. Work sponsored by the Office of Naval Research in Undersea Signal Processing.

9:55–10:15 Break

10:15

2aSP6. Coherent processing of shipping noise for ocean monitoring. Shane W. Lani (School of Mech. Eng., Georgia Institute of Technology, Atlanta, GA 30332, lani.shane@gatech.edu), Karim G. Sabra (School of Mech. Eng., Georgia Institute of Technology, Atlanta, GA), Philippe Roux (Institut des Sciences de la Terre, Université Joseph Fourier, Grenoble, None, France), William Kuperman, and William Hodgkiss (University of California, San Diego, CA)

Extracting coherent wavefronts between passive receivers using cross-correlations of ambient noise may provide a means for ocean monitoring without conventional active sources. Hence applying this technique to continuous ambient noise recordings provided by existing or future ocean observing systems may contribute to the development of long-term ocean monitoring applications such as passive acoustic thermometry. To this end, we investigated the emergence rate of coherent wavefronts over 6 days using low-frequency ambient noise ($f < 1.5$ kHz) recorded on two vertical line arrays-separated by 500m- deployed off San-Diego CA in ~200m deep water. The recorded ambient noise was dominated by nonstationary distributed shipping activity with

the frequent occurrence of loud isolated ships. Noise data were first processed to mitigate the influence of these loud shipping events in order to primarily emphasize the more homogenous and continuous background ambient noise in the frequency band. Furthermore, the coherent noise field propagating between the VLAs was beamformed using spatio-temporal filters to enhance the emergence rate of specific coherent wavefronts. This presentation will discuss various strategies for the selection of these spatio-temporal filters (either data-derived or model-based) in order to improve the continuous tracking of these coherent wavefronts over 6 days.

10:30

2aSP7. Estimation of a broadband response with dilation process compensation at very low signal to noise ratios. Paul J. Gendron (Maritime Systems Division, SSC Pacific, A460, Bldg. 1, Bayside Campus, 53560 Hull St., San Diego, CA 92152, paul.gendron@navy.mil)

Challenges of estimating broadband acoustic response functions at low signal to noise ratio (SNR) are due to both their varying sparsity and the varying spatio-temporal dynamics of each acoustic arrival. Acoustic responses can be quite sparse over the delay-Doppler-angle domain exhibiting large regions that are relatively quiet. The arrivals may share significant Doppler processes due to platform motion or may be driven independently by boundary interactions. Because of this estimation must be adaptive across delay-Doppler and angle with any single fixed estimator inadequate. One means of constructing such an estimator is to view each angle-Doppler-frequency slot as either ensonified or not. A mixture model can be employed for this purpose to describe the behavior of the acoustic response over received signal duration, aperture, and bandwidth. The posterior mean is derived and shown to be soft shrinkage operator of the conventional Wiener filtered coefficients under each of the components of the mixture. This estimator can be employed for bulk dilation estimation as an alternative to a phase locked loop. The posterior variance is derived and compared conventional Wiener filtering. The resulting adaptive structure is applied to M-ary orthogonal signaling sets taken in diverse shallow water environments at very low SNR. This work was supported by the Naval Innovative Science and Engineering Program and the Office of Naval Research.

TUESDAY MORNING, 23 OCTOBER 2012 MARY LOU WILLIAMS A/B, 8:00 A.M. TO 11:45 A.M.

Session 2aUW

Underwater Acoustics and Acoustical Oceanography: Propagation Topics

Ralph A. Stephen, Chair
Woods Hole Oceanographic Institution, Woods Hole, MA 02543-1592

Contributed Papers

8:00

2aUW1. Nonlinear acoustic pulse propagation in range-dependent underwater environments. Joseph T. Maestas (Mechanical Engineering, Colorado School of Mines, 1500 Illinois Street, Golden, CO 80401, jmaestas@mines.edu) and Jon M. Collis (Applied Mathematics and Statistics, Colorado School of Mines, Golden, CO)

The nonlinear progressive wave equation (NPE) is a time-domain formulation of Euler's fluid equations designed to model low angle wave propagation using a wave-following computational domain [B. E. McDonald et al., JASA 81]. The wave-following frame of reference permits the simulation of long-range propagation that is useful in modeling the effects of blast waves in the ocean waveguide. The standard formulation consists of four separate mathematical quantities that physically represent refraction, nonlinear steepening, radial spreading, and diffraction. The latter two of these effects are linear whereas the steepening and refraction are nonlinear. This formulation recasts pressure, density, and velocity into a single variable, a dimensionless pressure perturbation, which allows for greater efficiency in calculations. Nonlinear effects such as weak shock formation are accurately captured with the NPE. The numerical implementation is a combination of two numerical schemes: a finite-difference Crank-Nicholson algorithm for the linear terms of the NPE and a flux-corrected transport algorithm for the nonlinear terms. While robust, solutions are not available for sloping seafloors. In this work, range-dependent environments, characterized by sloping bathymetry, are investigated and benchmarked using a rotated coordinate system approach.

8:15

2aUW2. A comprehensive study of the Bellhop algorithm for underwater acoustic channel modelings. Xiaopeng Huang (Dept. of Electrical and Computer Engineering, Stevens Institute of Technology, Castle Point on Hudson, Hoboken, NJ 07030, xiaopeng.huang0508@gmail.com)

Ray tracing is one of the most conventional methods for modeling underwater acoustic sound propagation, and the Bellhop algorithm is a highly effcient ray tracing program, written by Michael Porter as part of the Acoustic Toolbox. In this abstract, based on the introduced Bellhop algorithm, we select several typical underwater acoustic environments so as to study how to model their channels and analyze their channel properties. Simulation results will investigate the following aspects of channel modeling and perperties: ray tracing, eigen-ray tracing, coherent transmission loss, channel impulse reposne, coherence time. In addition, simulation results will compare the performance difference with variant environmental parameters, such as sound speed anomaly and wavy surface.

8:30

2aUW3. Information content of an acoustic field propagating in an ocean waveguide. Steven Finette (Acoustics Division, Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

It is intuitively clear that, in some sense, waves carry "information" concerning both their source characteristics and their interaction with boundaries and/or sound speed inhomogeneity in the propagation path. This presentation addresses the issue of how one can estimate the maximum rate that an acoustic field can transfer information in an ocean waveguide, based on the properties of wave propagation. Information theory is the natural framework for addressing this question, relating wave propagation and communication concepts; it is applied here for the example of a Pekeris waveguide. Using properties of the propagation operator, information-theoretic arguments applied to the propagated field allow for the transfer of information along independent communication channels in the waveguide and an explicit expression for the channel capacity is obtained. The latter represents an upper bound on the error-free transfer of information from a source to a point in the waveguide by use of the propagated field. Work supported by the Office of Naval Research.

8:45

2aUW4. Empirical and collocation-point methods for estimation of acoustic field and array response probability density functions. Thomas J. Hayward and Roger M. Oba (Naval Research Laboratory, 4555 Overlook Ave SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Recent research has investigated the representation of acoustic field uncertainties arising from uncertainties of the acoustic environment, with emphasis placed either on the representation of the acoustic field as a random process or on estimation of the acoustic field probability density function (pdf) at a given receiver location. The present work introduces two methods for estimating acoustic field and array response pdfs. The first method is based on the computation of the empirical characteristic function (ECF) of the acoustic field, derived from a random sample in the acoustic parameter space and computation of the corresponding acoustic fields. The acoustic field pdf estimate is obtained as the Fourier transform of the ECF. The second method is based on approximation of the characteristic function using collocation-point methods, which are based on orthogonal-polynomial approximations of the mapping from the parameter space to the acoustic field. Both methods are investigated in two examples: (1) a stratified shallow-water model with two-dimensional uncertainties of sound speed and attenuation coefficient; and (2) a shallow-water model with sound-speed fluctuations in the water column defined by a time-stationary internal wave field. Sampling requirements and convergence of the pdf estimates are investigated for both methods and compared. [Work supported by ONR.]

9:00

2aUW5. Modeling uncertain source depth in range-dependent environments. Kevin R. James and David R. Dowling (Mechanical Engineering, University of Michigan, Mechanical Engineering, Ann Arbor, MI 48105, drd@umich.edu)

Efficient and accurate estimation of the uncertainty in a transmission loss calculation is important for tactical applications of underwater acoustic propagation calculations. Uncertainty in source depth can contribute significantly to the overall transmission loss uncertainty. The unique relationship between source depth and transmission loss motivates a different approach to uncertainty estimation than that used for other environmental and sound-channel parameters. Prior research has shown that in a range-independent environment, source depth uncertainty can be efficiently modeled using the principles of reciprocity. This presentation describes a new approach to uncertainty estimation in range-dependent environments, based on the assumption that the relationship between source depth and transmission loss is approximately governed by the adiabatic approximation on a local scale. Transmission loss predictions are taken from RAMGEO results to solve for the unknowns in the resulting approximate formulation. By modeling the relationship between source depth and transmission loss, approximate uncertainty bounds can be generated for transmission loss predictions. Results are provided for simple up-sloping and down-sloping range-dependent environments, for frequencies from 100 Hz to several kHz, and for ranges of several kilometers. [Sponsored by the Office of Naval Research, Code 322OA.]

9:15

2aUW6. Mode coupling due to bathymetric variation. Charles E. White, Cathy A. Clark (Naval Undersea Warfare Center, 1176 Howell Street, Newport, RI 02841, charlie.e.white@navy.mil), and Gopu Potty (Ocean Engineering, University of Rhode Island, Narragansett, RI)

In shallow water the assumption of range independence fails in conditions of rapidly-varying bathymetry and/or horizontal sound speed. In these environments, the modes of vibration of the acoustic wave equation become coupled, with a transfer of energy between adjacent modes occurring upon traversing a horizontal change of environment. In this talk, we will consider some simple applications of mode conversions due to variable bathymetry. Results will be compared to closed form propagation solutions in constant-slope wedge environments. The ultimate goal of this research is the development of a fully non-adiabatic range-dependent mode solution which retains analytical integrity while executing in a time window that is tactically useful for warfare applications.

9:30

2aUW7. Transport theory applied to shallow water acoustics: The relative importance of surface scattering and linear internal waves. Kaustubha Raghukumar and John A. Colosi (Naval Postgraduate School, 833 Dyer Rd, Monterey, CA 93943, kraghuku@nps.edu)

Acoustic fields in shallow water have a statistical nature due to complex, time-evolving sound speed fields and scattering from rough boundaries. A coupled-mode transport theory [Creamer (1996), Colosi and Morozov (2009)] allows for the prediction of acoustic field second moments like mean intensity and coherence. This was previously applied to study low frequency acoustic fluctuations in an environment typical of that of the Shallow Water 2006 (SW06) experiment on the New Jersey Continental shelf. Here the propagation was found to be strongly adiabatic and random sound speed fluctuations from internal waves radically altered acoustic interactions with intense nonlinear internal wave packets. Here, we extend the SW06 study to examine the ability of transport equations to describe high frequency (>1 kHz) sound in shallow water. Mode coupling rates from internal waves are expected to be larger, and scattering effects from rough surfaces need treatment. The aforementioned transport theory is merged with the rough surface scattering transport theory of Thorsos et al (2009). Oceanographic and sea surface measurements are used to constrain the internal wave and sea surface models. The relative importance of linear internal waves and surface scattering effects are studied using transport theory and Monte Carlo simulations.

9:45

2aUW8. Theory of the sound field fluctuations in the presence of internal waves due to adiabatic mechanism of interaction. Boris Katsnelson (Dept. of Physics, Voronezh State Univ., 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru), Mohsen Badiey (College of Earth, Ocean and Environment, University of Delaware, Newark, DE), and Alexander Tckhoidze (Dept. of Physics, Voronezh State Univ., Haifa, Haifa, Israel)

In the presence of moving nonlinear internal waves character of interaction between sound field and internal waves depends on angle between direction of an acoustic track and wave front of internal waves (mode coupling, horizontal refraction or adiabatic regime) [JASA, vol. (122), pp. 747-760, 2007]. In particular, if this angle is about 15-20 degrees there should be adiabatic mechanism. Adiabatic regime of propagation means that variations of the sound field follow variations of the sound speed profile. In this paper similar situation is considered when wave front of the train of internal waves crosses the acoustic track at the angle about 15 degrees. Theoretical modeling shows specific features of adiabatic fluctuations: variations of shape of waveguide modes, fluctuations of amplitudes (excitation coefficients) of the corresponding modes and phase fluctuations. These fluctuations can be separated in time on dependence on position of the train. Results of modeling are compared with experimental data [shown in accompanying paper, Badiey et al.] and are in good agreement. This work was supported by ONR and RFBR.

10:00–10:15 Break

10:15

2aUW9. Acoustic frequency shifts due to internal tides and nonlinear internal waves. Altan Turgut and Peter C. Mignerey (Acoustics Div., Naval Research Lab, Acoustics Div., Washington, DC 20375, altan.turgut@nrl.navy.mil)

Significant frequency shifts of acoustic intensity level curves in broadband signal spectrograms were measured in the East China Sea during the summer of 2008. Broadband pulses at 270-330 Hz were transmitted from a fixed source and received at a bottomed horizontal array, located at 33 km range. The acoustic intensity level curves of the received signals indicate regular frequency shifts that are well correlated with the measured internal

2a TUE. AM

tides and nonlinear internal waves. Regular frequency shifts due to nonlinear internal waves are observed only when their wave-fronts are nearly parallel to the acoustic propagation path, causing an effective change in the sound speed profile. Similar effects were observed in 3-D numerical simulation results when curved nonlinear internal wave fronts are used. These observations and simulations indicate the potential of monitoring internal tides and nonlinear internal waves using low-frequency acoustic signals when the acoustic source and receiver are strategically placed. [Work supported by the Office of Naval Research.]

10:30

2aUW10. Range dependent acoustic intensity scintillations due to focusing, defocusing, and scattering by sea swell and bottom sediment waves. Alexey A. Shmelev (WesternGeco, Schlumberger, Houston, TX 77057, alexey.a.shmelev@gmail.com), James F. Lynch, Ying-Tsong Lin, Arthur E. Newhall, and Timothy F. Duda (AOPE, Woods Hole Oceanographic Institution, Woods Hole, MA)

It is known that the waveguide depth variability causes horizontal refraction and coupling of acoustic normal modes. Presence of large bottom sediment waves and sea swell are examples of strongly anisotropic waveguides that result in range dependence of the acoustic scintillation index. In the directions parallel to the wave crests, three-dimensional effects of mutual horizontal focusing, defocusing and diffusion between such waves are the main mechanisms of intensity fluctuations. For acoustic propagation in the perpendicular to the wave crests directions, intensity fluctuations are mainly driven by random mode coupling and scattering. Analytical studies and numerical examples of the acoustic scintillation index, as well as its azimuthal and range dependence in the shallow water with both types of waves, will be provided. Directions for future studies will be discussed.

10:45

2aUW11. Deep seafloor arrivals in long range ocean acoustic propagation. Ralph A. Stephen, S. Thompson Bolmer, Matthew A. Dzieciuch (Geol & Geophys, WHOI, 360 Woods Hole Rd, Woods Hole, MA 02543-1592, rstephen@whoi.edu), Peter F. Worcester (IGPP, Scripps Institution of Oceanography, La Jolla, CA), Rex K. Andrew, James A. Mercer (Applied Physics Laboratory, University of Washington, Seattle, WA), John A. Colosi (Oceanography, Naval Postgraduate School, Seattle, WA), and Bruce M. Howe (Ocean and Resources Engineering, University of Hawaii at Manoa, Honolulu, HI)

Ocean bottom seismometer observations during the long-range ocean acoustic propagation experiment in the North Pacific in 2004 showed robust, coherent, late arrivals that were not observed on hydrophones suspended 750m and more above the seafloor and that were not readily explained by ocean acoustic propagation models. The DSFA arrival pattern on the OBSs near 5000m depth are a delayed replica, by about two seconds, of the arrival pattern on the deepest element of the DVLA at 4250m depth (DVLA-4250). Using a conversion factor from the seafloor vertical particle velocity to seafloor acoustic pressure, we have quantitatively compared signal and noise levels at the OBSs and DVLA-4250. Ambient noise and DSFA signal levels at the OBSs are so quiet that if the DSFA arrivals were propagating through the water column, perhaps on an out-of-plane bottom-diffracted-surface-reflected (BDSR) path, they would not appear on single, unprocessed DVLA channels. Nonetheless arrival time and horizontal phase velocity analysis rules out BDSR paths as a mechanism for DSFAs. Whatever the mechanism, the measured DSFAs demonstrate that acoustic signals and noise from distant sources can appear with significant strength on the seafloor at depths well below the conjugate depth.

11:00

2aUW12. Effects of fine-scale topographical change on mid-frequency bottom loss. Jie Yang (Applied Physics Lab, University of Washington, 1013 NE 40th St, Seattle, WA 98105, jieyang@apl.washington.edu) and Dajun Tang (Applied Physics Lab, University of Washington, Seattle, WA)

It was shown previously [Yang et al, J. Acoust. Soc. Am. 131(2), 1711-1721 (2012)] that forward scatter from topographical changes could alter bottom loss at mid-frequencies. In this more detailed study, fine-scale bottom bathymetry data from a multibeam survey are used as inputs to numerical experiments to investigate the effects of topographical variation on bottom loss (BL). Bottom reflection/forward scatter simulations in the frequency band of 2–5 kHz are carried out using several numerical methods which all include the effect of bathymetry variation. Bottom bounces including forward scatter are treated as if the bottom is flat and are used to estimate BL at different frequencies. It is found that small topographic changes can result in large deviations in BL estimates. Remedies for the effect of bottom topography change on BL are suggested. [Work supported by ONR.]

11:15

2aUW13. Modeling the effect of interface roughness on bottom loss from layered interfaces with finite elements. Marcia J. Isakson and Nicholas P. Chotiros (Applied Research Laboratories, University of Texas, 10000 Burnet Road, Austin, TX 78713, misakson@arlut.utexas.edu)

The bottom loss from a layered ocean sediment is determined using a finite element/boundary element (FE/BE) method. First, the pressure and its normal derivative are calculated on the top interface using finite elements. Then the field at a point outside of the domain is determined using the Helmholtz/Kirchhoff integral (BE). Bottom loss is then calculated by comparing the reflected/scattered energy to the incident energy. The finite element method makes no approximations to the Helmholtz equation and is exact within the limits of the discretization. Any number of layers including elastic layers with rough or smooth interfaces can be included. The results of the FE/BE approach will be compared to Geoacoustic Bottom Interaction Model (GABIM) for a number of test cases. [Jackson, et al., IEEE J. Ocean. Eng. 35(3), 603-617 (2010)] GABIM computes the layered reflection coefficient then includes scattering for one rough interface based on "a combination of the Kirchhoff approximation, first-order perturbation theory and an empirical expression for very rough seafloors". Lastly, the bottom loss of multiple rough interfaces will be compared to that of a single rough interface. [Work supported by ONR, Ocean Acoustics.]

11:30

2aUW14. Jurassic acoustics: Low frequency sound absorption in the ocean during past ages. David Browning (Physics Department, URI, Kingston, RI 02881, decibeldb@aol.com) and Peter M. Scheifele (Communication Sciences Dept., University of Cincinnati, Cincinnati, OH)

A major aspect of global warming is ocean acidification. To provide a baseline for future change, investigators have been able to track the geological record of ocean acidification back to 300 million years ago (mya). One of the key factors is tracing the history of boron isotopes. The principal low frequency sound absorption mechanism in seawater is a boron reaction which is pH dependent (the lower the pH, the lower the absorption), so this geological record can be used to estimate sound absorption in the ocean all the way back to the carboniferus period. The broad picture is that low frequency absorption in the ocean decreased from 300 mya to 200 mya, was relatively constant from 200 mya to 100 mya, and then has been increasing since. The present level is back to one similar to that 300 mya. Future global warming may reverse this trend and cause the absorption to decrease down to a level similar to when the dinosaurs roamed (100 mya).

Meeting of the Standards Committee Plenary Group
to be held jointly with the meetings of the
ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:
ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 43/SC 3, Underwater acoustics
ISO/TC 108, Mechanical vibration, shock and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures,
ISO/TC 108/SC 3, Use and calibration of vibration and shock measuring instruments,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machines,
and
IEC/TC 29, Electroacoustics

P.D. Schomer, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43 Acoustics and ISO/TC 43/SC 1 Noise
Schomer and Associates, 2117 Robert Drive, Champaign, Illinois 61821

M.A. Bahtiarian, Acting Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 43/SC 3 Underwater acoustics
Noise Control Engineering, Inc., 799 Middlesex Turnpike, Billerica, MA 01821

D.J. Evans, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108 Mechanical vibration, shock and condition monitoring, and ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring devices
National Institute of Standards and Technology (NIST), 100 Bureau Drive, Stop 8220, Gaithersburg, MD 20899

W.C. Foiles, Co-Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
BP America, 501 Westlake Park Boulevard, Houston, TX 77079

R. Taddeo, Co-Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
NAVSEA, 1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington, DC 20376

D.D. Reynolds, Chair, U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
3939 Briar Crest Court, Las Vegas, Nevada 89120

D.J. Vendittis, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
701 Northeast Harbour Terrace, Boca Raton, FL 33431

R. Taddeo, Vice Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
NAVSEA, 1333 Isaac Hull Avenue, SE, Washington Navy Yard, Washington, DC 20376

V. Nedzelnitsky, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
National Institute of Standards and Technology (NIST), Sound Building, Room A147, 100 Bureau Drive, Stop 8221, Gaithersburg, MD 20899-8221

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting.

The meeting of the Standards Committee Plenary Group will precede the meetings of the Accredited Standards Committees S1, S2, S3, S3/SC 1, and S12, which are scheduled to take place in the following sequence:

Tuesday, 23 October 2012	10:30 a.m.–11:30 a.m.	ASC S12, Noise
Tuesday, 23 October 2012	1:15 p.m.–2:15 p.m.	ASC S1, Acoustics
Tuesday, 23 October 2012	2:30 p.m.–3:45 p.m.	ASC S3, Bioacoustics
Tuesday, 23 October 2012	4:00 p.m.–5:00 p.m.	ASC S3/SC 1, Animal Bioacoustics
Wednesday, 24 October 2012	8:30 a.m.–9:45 a.m.	ASC S2, Mechanical Vibration & Shock

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3 and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
P.D. Schomer, Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
P.D. Schomer, Chair	**ISO/TC 43/SC 1** Noise	ASC S12
M.A. Bahtiarian, Acting Chair	**ISO/TC 43/SC 3** Underwater acoustics	ASC S1, ASC S3/SC 1 and ASC S12
D.J. Evans, Chair	**ISO/TC 108** Mechanical vibration, shock and condition monitoring	ASC S2
W.C. Foiles, Co-Chair R. Taddeo, Co-Chair	**ISO/TC 108/SC 2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures	ASC S2
D.J. Evans, Chair	**ISO/TC 108/SC 3** Use and calibration of vibration and shock measuring instruments	ASC S2
D.D. Reynolds, Chair	**ISO/TC 108/SC 4** Human exposure to mechanical vibration and shock	ASC S3
D.J. Vendittis, Chair R. Taddeo, Vice Chair	**ISO/TC 108/SC 5** Condition monitoring and diagnostics of machines	ASC S2
IEC		
V. Nedzelnitsky, U.S. TA	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

TUESDAY MORNING, 23 OCTOBER 2012 TRIANON E, 10:30 A.M. TO 11:30 A.M.

Meeting of Accredited Standards Committee (ASC) S12 Noise

W. J. Murphy, Chair, ASC S12
NIOSH, 4676 Columbia Parkway, Mail Stop C27, Cincinnati, OH 45226

S.J. Lind, Vice Chair, ASC S12
The Trane Co., 3600 Pammel Creek Road, Bldg. 12-1, La Crosse WI 54601-7599

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note—that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 23 October 2012.

Scope of S12: Standards, specifications and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation and control, including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

Session 2pAA

Architectural Acoustics: Coordination of Architectural and Sound System Design in the Built Environment

Kenneth Roy, Cochair
Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604

Joel A. Lewitz, Cochair
Rosen Goldberg Der & Lewitz, 1100 Larkspur Landing Cir., Larkspur, CA 94939

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. Case studies of a method to integrate architectural acoustic and sound systems design. Gary W. Siebein and Hyun Paek (Siebein Associates, Inc., 625 NW 60th Street, Suite C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com)

A method to evaluate the integrated architectural acoustic and reinforced sound systems for medium to large size worship and performance spaces using computer modeling of the systems for new facilities and diagnostic impulse response based measurements in existing rooms is presented. Four case studies of varying degrees of design integration of architectural acoustic and sound systems are presented to illustrate the method. Case study 1 is a divisible multi-purpose worship space. The sound system equipment and design was done without considering the acoustical design of the room. It was decided to re-use the existing equipment and adjust the aiming and programming of the system to fit the acoustics and architecture of the room to optimize the balance of natural and reinforced sounds. The second case study covers the design of a theater with a distributed array system to reinforce theatrical and musical sounds on stage. The third case study covers the installation of a new sound system in a large worship space with an organ and a very long reverberation time. The fourth case study covers the diagnostics, architectural and sound system improvements in a unique space for symposia on a college campus.

1:25

2pAA2. Naturally you need a sound system. Rogers Dixson (Cape Dixson Associates, Inc., 4279 Roswell Rd., NE, Ste 102-135, Atlanta, GA 30342, rdixson@cdai.com)

Ignoring Room Acoustics in Audio System design can be just as problematic as ignoring the audio system in designing the acoustics of a space. Contrary to marketing claims by product manufacturers on either side of the design "aisle," neither system can overcome all of the shortcomings imposed by the poor design of the other. Both elements have to be designed in coordination with each other (as well as in coordination with other systems in the built environment). This can become particularly challenging in multipurpose performance and presentation spaces as there can be multiple audio systems installed in the same space. Sound Reinforcement, Program Playback (particularly multi-channel surround sound) systems are increasingly being installed in performing arts facilities where natural acoustics is a critical element. How these aspects should (and can) be coordinated is discussed in this paper.

Contributed Papers

1:45

2pAA3. Seamless integration of audio visual design into architecture for more successful projects. Felicia Doggett (Metropolitan Acoustics, LLC, 40 W. Evergreen Ave., Suite 108, Philadelphia, PA 19118, felicia@metropolitanacoustics.com)

The integration of audio visual systems into technology-enabled buildings has long been a struggle with the architectural community. In some instances, architects do not want any part of the systems to be visible within the space, like speakers, projectors and equipment racks. In other cases, like digital signage and way-finding technology, the systems are front and center in the room and must be as visible and as user-accessible as possible. Audio visual technology is present in some form in almost every building being constructed today including educational, corporate, houses of worship, legislative, hospitality, sports arenas, healthcare, museums, and retail. Much time is spent not just on the system design, but working with architects on integrating the technology into these built environments, each of which has its own requirements. This presentation focuses on various ways to integrate audio visual technology into buildings for more seamless and successful projects. Balancing the functionality and performance of any audio visual system installation with the aesthetic impact on the space takes experience, creativeness, and a willingness to coordinate with the design team.

2:00

2pAA4. Sound system installed in a central bus station. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

As a result of a new administration board planning in the Central long range Bus Station located in a mayor city, it was installed a public address system, together with a small modification of the visible building materials within the station, for the purpose of easy understanding of; bus departure

confirmation, people and stuff localization and general information to passengers, workers and visitors of the station. The system involved a distributed sound system with dozens of loudspeakers strategically located with reasonable, non-disturbing, sound pressure level generation by each one, which enhanced the intelligibility of the spoken messages well over the average for the Central Bus Stations system in the country. The system performed at a comfortable sound level, just enough to overcome the noise level within the central, which together with a reduced reverberation time allowed for proper effortless understanding of the emitted messages. Mayor contributors for the obtained results were careful design, sound equipment quality, sound absorption materials, guided installation and luck.

2:15

2pAA5. Active acoustics in a restaurant: A case study. Pierre Germain and Roger W. Schwenke (Meyer Sound Laboratories, 2832 San Pablo Ave, Berkeley, CA 94702, rogers@meyersound.com)

Loud restaurants have become sufficiently common that newspapers have begun rating the sound levels of restaurants. A Zagat survey in 2011 found that diners ranked noise as their second-highest complaint behind service. Comal Restaurant in Berkeley California is the first to use a combination of active and passive acoustic treatments to control the reverberation time of the dining area.

2:30

2pAA6. Sight and sound: Visual aesthetics of loudspeakers. Ben Bridgewater, Bob Coffeen, and Jim Long (University of Kansas, Lawrence, KS 59741, BenBridgewater@gmail.com)

Loudspeakers are not only heard but often seen. A sound system's loudspeakers must be designed to meet the visual expectations of the architect, performers, and owners. A good designer can meet the audio requirements while not offending the architecture of the space. A case study of hidden, obvious, and ugly loudspeakers will be presented.

2:45

2pAA7. Review of an arena's acoustical and electroacoustical design upgrades. David Scheirman (Harman Professional, 8500 Balboa Blvd, Northridge, CA 91329, dscheirm@harman.com)

Sports arenas and stadiums are an important economic part of the urban centers in which they are located. For cost efficiency, acoustical elements of the original architectural design should complement and support the facility's requirements for communication and public address equipment. Multi-purpose use requirements are examined for one such facility, an indoor arena which measures 950,000 square feet (88,257.9 m^2) of total space, with a 94-foot (28.7 m) by 200-foot (61.0 m) arena floor. It hosts over 250 events and nearly 4,000,000 guests each year, seating 18,000-20,000 attendees per event based upon type and format. The timeline is reviewed over which acoustical and electroacoustical studies, upgrades and modifications were carried out. Original-construction sprayed cellulose acoustical insulation was enhanced by installation of lapendary panels to reduce reverberation time and sound intensity levels in target frequency bands. Finally, with the installation of a new multi-array line-array type loudspeaker system, the venue has achieved increased speech intelligibility and improved audience-perceived quality of sound reproduction for both sports and entertainment production events. The upgrades have helped to enhance its global reputation as a multi-functional space. The building was voted Venue of the Year at the 2011 Stadium Business Awards.

TUESDAY AFTERNOON, 23 OCTOBER 2012 — JULIA LEE A/B, 1:00 P.M. TO 5:45 P.M.

Session 2pAB

Animal Bioacoustics: Arctic Bioacoustics

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

Chair's Introduction—1:00

Invited Papers

1:05

2pAB1. Soundscape of the North-Eastern Chukchi Sea. Bruce Martin (JASCO Applied Sciences, 32 Troop Avenue, Suite 202, Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), Julien Delarue, and David Hannay (JASCO Applied Sciences, Victoria, British Columbia, Canada)

The Chukchi Sea is a dynamic environment that cycles each year from open seas in summer to 100% ice cover in winter. The ice and sea conditions lead to a highly variable acoustic background. In summer the soundscape's backdrop are wind and wave sounds typical of shallow seas. In the winter, grinding ice sheets can create a cacophony of unusual sounds, some of which can easily be mistaken as being biological in origin. During ice formation and break-up sound levels can increase significantly due to combinations of wind and waves and ice floe collisions. Several marine mammal species move through this environment and their calls sometimes dominate the soundscape. Bearded seals are year-round residents whose calls overwhelm the soundscape in the spring during their mating season. Walrus follow the ice edge and fill their neighbourhoods with grunts, moans and knocks. Seasonal migrants such as bowhead and beluga whales are heard passing through in spring and fall. Summer visitors such as fin, killer, minke and gray whales add to the voices of this environment. Anthropogenic sounds mix into the soundscape during the open water season and are concentrated along the coast lines and in areas of interest for oil and gas exploration. This presentation will provide examples of all these sounds and will show spatial-temporal maps of the sounds distributions based on data from the Chukchi autonomous recording array operated by the Joint Studies Program since 2007.

1:25

2pAB2. Seals and sound in a changing Arctic: Ongoing psychoacoustic studies of spotted and ringed seals. Jillian Vitacco (Ocean Sciences, University of California at Santa Cruz, 100 Shaffer Rd, Santa Cruz, CA 95050, jillian.vitacco@gmail.com), Colleen Reichmuth, Asila Ghoul (Long Marine Laboratory, Institute of Marine Sciences, University of California, Santa Cruz, CA), and Brandon Southall (Southall Environmental Associates, Aptos, CA)

Arctic environments are changing rapidly as a result of climate warming and industrialization, and as sea ice recedes, activity associated with transportation and oil and gas production increases. Among the many concerns for ice-living seals in the region is the potential for behavioral or auditory effects resulting from noise exposure. Currently there are limited data available concerning the hearing sensitivity of arctic seals - some data exist for harp and ringed seals, while the most comprehensive data are for harbor seals. As the phylogenetic relationships among northern seals are not well resolved, extrapolation across species for management purposes is difficult. To this end, we are working to describe the species-typical hearing of spotted (*Phoca largha*) and ringed seals (*Pusa hispida*). Thus far, measurements of the underwater hearing sensitivity of spotted seals show best sensitivity between 3.2–25.6 kHz and peak sensitivity of 51 dB re 1 μPa at 25.6 kHz. Absolute thresholds for airborne tonal signals indicate acute sensitivity of $<$10 dB re 20 μPa from 0.80–12.8 kHz. Audiometric testing for ringed seals is ongoing, as are critical ratio measurements for both species. These studies will provide valuable insight into how arctic seals perceive acoustic signals, as well as inform management practices for these vulnerable species.

1:45

2pAB3. Simultaneous sound production in the bowhead whale Balaena mysticetus—Sexual selection and song complexity. Outi M. Tervo (Arctic Station, University of Copenhagen, Post box 504, Qeqertarsuaq 3905, Greenland, outiter@gmail.com), Lee A. Miller (Institute of Biology, University of Southern Denmark, Odense, Denmark), and Mads F. Christoffersen (Arctic Station, University of Copenhagen, Qeqertarsuaq, Qaasuitsup, Greenland)

Different components of bowhead whale Balaena mysticetus song were localized using hydrophone arrays. In 2008 recordings were made using two hydrophones spaced 20–35 m apart. In 2009 a linear GPS synchronized array of four hydrophones with an aperture of ~1400 m was used. The localization results confirm the co-location of the sound sources. The analyses show amplitude modulation of one signal caused by the onset of the second signal, which provides additional evidence of simultaneous sound production. Sound, when used as an indicator of fitness forces the vocalizing animal to improve the quality of its signal to compete with other vocalizing conspecifics. Several methods can be used to improve signal quality and these include 1) large repertoire size, 2) annually/seasonally changing repertoire, 3) broad frequency band, and 4) simultaneous sound production. Bowhead whales show all these features in their acoustic behaviour and we suggest that these complex songs have evolved as the result of sexual selection. Song complexity has been shown to be of importance in the sexual selection of many song bird species implying that sound complexity may be a key factor in the sexual behaviour of bowhead whales.

2:05

2pAB4. Bowhead whales and airgun pulses: Detecting a threshold of behavioral reaction. Susanna B. Blackwell (Greeneridge Sciences, Inc., Santa Barbara, CA 93117, susanna@greeneridge.com), Trent L. McDonald, Christopher S. Nations (WEST, Inc., Cheyenne, WY), Aaron M. Thode (Marine Physical Laboratory, Scripps Institution of Oceanography, San Diego, CA), Katherine H. Kim, Charles R. Greene (Greeneridge Sciences, Inc., Santa Barbara, CA), and Michael A. Macrander (Shell Exploration and Production Co., Anchorage, AK)

Previous work has shown that bowhead whales cease calling when near ($<$40 km) seismic exploration activities involving airguns. The aim of this study is to estimate the received level threshold for the onset of this behavior (cessation of calling). The analysis relied on data collected during late summer of 2007-2010 by up to 40 Directional Autonomous Seafloor Acoustic Recorders (DASARs) in the Beaufort Sea. About 98,000 localized calls and hundreds of thousands of airgun pulses were included in the analysis. For each 10-min period of data collected at each recorder, each year, the cumulative sound exposure level (CSEL) from airgun pulses was calculated and paired with the number of calls concurrently localized within ~3.5 km of each DASAR. Poisson regression was then used to estimate the threshold of airgun sound exposure received at the whales when call cessation begins. The CSEL threshold was found to be near 124 dB re 1 μPa^2 s (95% confidence intervals = 119-129 dB). For an airgun array firing every 10 sec, this corresponds to a received single pulse SEL at the whale of ~106 dB re 1 μPa^2 s. [Work supported by Shell Exploration and Production Company.]

2:25

2pAB5. Arctic marine mammal passive monitoring and tracking with a single acoustic sensor. Juan I. Arvelo (Applied Physics Laboratory, Johns Hopkins University, 11100 Johns Hopkins Rd., Laurel, MD 20723, juan.arvelo@jhuapl.edu)

The Arctic Ocean is a unique environment in the number of physical mechanisms that may be potentially exploited with much simpler acoustic systems than would be required in other oceans. The Arctic sound speed profile forms a surface duct with favorable cylindrical-spreading for near-continuous detection of marine mammal vocalizations. This ducted waveguide exhibit low seasonal variability, particularly under the ice cap, forcing under-ice sound to heavily interact with this rough elastic stratified boundary. The ice roughness introduces steeper slopes that enhance water-to-ice sound penetration [Arvelo, POMA 2012]. The ice elasticity is responsible for the excitation of a radially polarized longitudinal wave and a transverse-horizontal shear wave with group velocities around 2700-3000 m/s and 1550-1650 m/s, respectively. A third dispersive flexural vertical plate wave propagates at much slower speeds ($<$1200 m/s) at low frequencies [Stein, Euerle & Parinella, JGR 1998]. Vocalization distance may be estimated from the time delays between the three wave types via blind deconvolution, while an arctangent bearing-estimator may increase the azimuthal localization resolution for high SNR vocalizations [Maranda, Oceans 2003]. Therefore, the unique Arctic environment is well suited for passive marine mammal monitoring and tracking with just a single ice-embedded geophone or under-ice vector sensor.

2:45–3:00 Break

3:00

2pAB6. Long-range tracking of bowhead whale calls using directional autonomous seafloor acoustic recorders. Delphine Mathias, Aaron M. Thode (Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, CA 92037-0238, delphine.mathias@gmail.com), Katherine H. Kim, Susanna B. Blackwell, Charles R. Greene (Greeneridge Sciences, Santa Barbara, CA), and Michael A. Macrander (Shell Exploration and Production Co., Anchorage, California)

Since 2007 "Directional Autonomous Seafloor Acoustic Recorders" (DASARs) have been deployed at five sites across a 280 km swath of the Beaufort Sea continental shelf to record bowhead whale (Balaena mysticetus) calls during their autumn migration. Composed of an omnidirectional pressure sensor and two horizontal directional sensors measuring particle motion, DASARs provide information for determining the bearing to a sound source. In previous analyses, bearings obtained from multiple DASARs within a single site have been used to localize calls, with a maximum baseline separation of 21 km between instruments. Here, we use data collected from two different sites to exploit a 45 km instrument separation for tracking bowhead whale calls detected under low ambient-noise situations in 2009 and 2011. These data sets have been manually analyzed to extract a series of calls from individual whales swimming from one site toward another. The fact that the tracking method does not require relative arrival time information makes matching calls between these widely separated shallow-water recordings practical. Both empirical and numerical transmission loss models are used to investigate the relationship between animal orientation, source level, and the presence of harmonics.

3:15

2pAB7. Acoustic monitoring of belugas (*Delphinapterus leucas*) in the eastern Chukchi Sea. Ellen C. Garland, Catherine Berchok (National Marine Mammal Lab, Alaska Fisheries Science Center, NOAA, AFSC/NOAA, 7600 Sand Point Way NE, Seattle, WA 98115, Ellen.Garland@noaa.gov), and Manuel Castellote (National Marine Mammal Lab, Alaska Fisheries Science Center, NOAA, Seattle, WA)

Beluga whales (*Delphinapterus leucas*) are highly vocal animals which make them ideal candidates for passive acoustic monitoring. Alaskan belugas overwinter in the Bering Sea, and in spring, two subpopulations migrate to their predictable summering grounds in the eastern Chukchi and eastern Beaufort Seas. In autumn, these subpopulations complete their annual migration by returning south to the Bering Sea. Additional information is required on the timing and migration routes in spring and autumn to assist in detecting these subpopulations as they transit between the two regions. Preliminary results are presented on the temporal distribution of Alaskan beluga based on acoustic detection (September 2010 to May 2011) from a passive acoustic recorder located 30 miles off Icy Cape, AK, in the eastern Chukchi Sea. Belugas were sporadically detected in autumn from mid-September to the start of December, with a peak in detections in late November. In spring, belugas were detected from mid-April to the end of May, with peaks in early and late May. Within each temporal peak in vocal activity we will investigate the common call types to assist in differentiating migratory streams, and potentially identify each subpopulation as they transit through this inshore area. [Work supported by the National Research Council and Bureau of Ocean Energy Management.]

3:30

2pAB8. Pacific walrus vocal repertoire in the northeastern Chukchi Sea: Call type description and relative proportion in time and space. Xavier Mouy (JASCO Applied Sciences, Victoria, BC, Canada), Julien Delarue (JASCO Applied Sciences, 202 - 32 Troop Ave, Dartmouth, NS B3B 1Z1, Canada, julien.delarue@jasco.com), Bruce Martin (JASCO Applied Sciences, Dartmouth, NS, Canada), and David Hannay (JASCO Applied Sciences, Victoria, BC, Canada)

Pacific walrus are present in the northeastern Chukchi Sea (NCS) from June to October. The study of their sounds has been largely restricted to the knock and bell sounds produced by males during the breeding season and in-air grunts and barks from mother and pups. A passive acoustic monitoring program conducted in the NCS since July 2006 has brought strong evidence that the underwater vocal repertoire of walrus is more diverse. Nine call types (including knocks and bells) and their variants identified over four years of acoustic monitoring will be described. Spectral measurements along with estimates of variability for high signal-to-noise ratio calls will be provided. The relative proportion of each call type across the study area and throughout the season is currently analyzed based on the identification of all calls in samples recorded multiple times per day in 2009 and 2010. Preliminary results suggest that the vocal repertoire of walrus is dominated by grunt-type calls, which is consistent with the NCS herds being mainly composed of females, pups and juveniles. The recurrent presence of knocking sounds indicates that either adult males routinely occur in the study area or that other age and sex classes also produce this call type.

3:45

2pAB9. Right whale versus bowhead whale gunshot calls in the Bering Sea. Catherine L. Berchok, Jessica L. Crance, Jessica L. Thompson, Stephanie L. Grassia, Phillip J. Clapham (National Marine Mammal Lab, NOAA/AFSC, 7600 Sand Point Way NE, Seattle, WA 98115, Catherine.Berchok@noaa.gov), Dana L. Wright (School of Fisheries and Ocean Sciences, University of Alaska Fairbanks, Fairbanks, AK), and Marc O. Lammers (Hawaii Institute of Marine Biology, University of Hawaii, Kaneohe, HI)

The Bering Sea is an important area for many cetaceans. It functions as both summer feeding grounds for some species like humpback and right whales, and wintering grounds for more northern species such as bowhead and beluga whales. Passive acoustics is an important tool for monitoring the presence of these and other species in the Bering Sea, although similarities in call characteristics among species can create confusion. For example, gunshot calls are produced by both right and bowhead whales. Recordings made in the Bering Sea (2007-2011) have been analyzed for the presence of gunshot calls. The spatio-temporal distribution of these calls is compared to that of more 'standard' bowhead and right whale calls. With the inclusion of information on historical range, migration patterns, and ice coverage, a tentative separation of the two species is proposed. Call characteristics and contextual information from the resulting subsets of data are then examined and compared with data confidently attributed to each species. Whether right and bowhead whale gunshot calls can be discriminated with sufficient reliability to include in spatio-temporal distribution analyses, and what added value this call type gives over using only 'standard' call types, will be discussed.

4:00

2pAB10. Cetacean vocalizations and anthropogenic noise levels in polar waters of the Atlantic. Sharon L. Nieukirk, Holger Klinck, Karolin Klinck, David K. Mellinger, Robert P. Dziak, and Haruyoshi Matsumoto (Cooperative Institute for Marine Resources Studies, Oregon State University and NOAA Pacific Marine Environmental Laboratory, 2030 SE Marine Science Drive, Newport, OR 97365, sharon.nieukirk@oregonstate.edu)

Obtaining baseline information on the distribution of endangered species in polar waters is important as climate change may adversely affect this fragile environment. In 2009 we began an acoustic survey in the Greenland Sea and Fram Strait to monitor the low-frequency calls from marine mammals using these waters. We also documented sources and levels of ambient noise as these will change as human use of the area increases. We recorded the vocalizations of numerous marine mammals; here we report our results for fin (Balaenoptera physalus), blue (B. musculus), sei (B. borealis) and sperm (Physeter macrocephalus) whales. The 20 Hz pulses of fin whales were recorded in the fall and early winter months. Sounds from blue whales, sperm whales and sei whales were recorded primarily in the summer and early fall. Background noise levels were dominated by the sounds from seismic airguns during the spring, summer and fall; during summer these sounds were recorded in all hours of the day and in all days of a month. Future increases in oil exploration and ship traffic coincident with melting sea ice will increase ambient noise levels, potentially affecting the numerous species of vocalizing whales using this area.

4:15

2pAB11. Spatio-temporal distribution of ice seals in the Chukchi Sea using underwater vocalizations: Special focus on male bearded seals. Heloise Frouin, Xavier Mouy, Julien Delarue (JASCO Applied Sciences, 2305 - 4464 Markham Street, Victoria, BC V8Z 7X8, Canada, heloise.frouin-mouy@jasco.com), Bruce Martin (JASCO Applied Sciences, Dartmouth, NS, Canada), and David Hannay (JASCO Applied Sciences, Victoria, BC, Canada)

Underwater vocalizations of ringed, ribbon and bearded seals were recorded over a wide region of the northeastern Chukchi Sea between July 2007 and October 2011. Ringed seals were identified by their barks and yelps, ribbon seals by their sweeps and puffing sounds and bearded seals by their trills, ascents and moans. Ringed seal vocalizations were detected in all months of the year, whereas vocalizations from ribbon seals occurred only in October and November. To determine the seasonal variation in the frequency of occurrence of male bearded seal vocalizations throughout a year 20-min recordings were analyzed to determine call counts between 02:00 and 06:00 every 3 days. Bearded seal acoustic detections increased progressively from August to March, peaked between April and June but were essentially absent in July. Outside of the mating period (April - June) the frequency of occurrence of vocalizations varied on a diel cycle and was higher during periods of darkness. To determine the influence of diel cycle, the duration of vocalizations and the proportion of each vocal type throughout the mating season 10-min recordings on a 17-20% duty cycle were analysed between April and June every 10 days. Results are currently under review and will be presented.

4:30

2pAB12. Spatio-temporal distribution of fin whales in the Bering Sea, 2007–2011. Jessica L. Thompson, Catherine L. Berchok, Phillip J. Clapham (National Marine Mammal Laboratory, NOAA/AFSC, 7600 Sand Point Way NE, Seattle, WA 98115, jessica.thompson@noaa.gov), Marc O. Lammers (Hawaii Institute of Marine Biology, University of Hawaii, Kaneohe, HI), and Sue E. Moore (National Marine Fisheries Service, Office of Science & Technology NOAA Fisheries, Seattle, WA)

The National Marine Mammal Laboratory has been collecting passive acoustic recordings of vocalizing marine mammals through much of the southeastern Bering Sea since 2000. The present analysis combines these recordings with those obtained in 2007 from a North Pacific Research Board-funded study (Stafford and Mellinger) to determine the long-term spatio-temporal distribution of fin whales throughout the Bering Sea shelf. A total of 28 moorings (19 Autonomous Underwater Recorders for Acoustic Listening (AURALs); 9 Ecological Acoustic Recorders (EARs)) were deployed year-round from October 2007 to September 2011 at varying depths and locations along the Bering Sea shelf, from Umnak Pass to just south of St. Lawrence Island. Preliminary analyses show an annual southward movement of calling fin whales in winter, seemingly associated with the expanding ice edge; these results will be compared with 1-day composite ice cover measured by the Advanced Microwave Scanning radiometer for the Earth Observing System (NOAA Coastwatch). Furthermore, the results show a considerably lower number of calls detected south of St. Matthews compared to the other mooring locations. In addition, overall fin whale seasonal calling trends along and between the 50m and 70m isobaths, and the species' use of Umnak Pass and Unimak Pass will be described.

4:45

2pAB13. Contributions of airgun activity to the ice-free ambient noise environment of the shallow Beaufort Sea between 2007 and 2011. Aaron Thode, Delphine Mathias, Katherine H. Kim, Susannna B. Blackwell (Scripps Institution of Oceanography, UCSD, 9500 Gilman Dr, La Jolla, CA 92093-0238, athode@ucsd.edu), Charles R. Greene (Greeneridge Sciences, Inc., Santa Barbara, CA), and Michael Macrander (Shell Exploration and Production Company, Anchorage, AK)

Every year since 2007 a collection of at least 35 "Directional Autonomous Seafloor Acoustic Recorders" (DASARs) have been deployed across a 280 km swath of the Beaufort Sea continental shelf, in water depths between 15 and 50 m. The ability of these instruments to estimate the arrival azimuth of transient signals has facilitated the development of an automated algorithm for the detection of airgun activity. This algorithm has been applied to five seasons of data, and in this presentation the contributions of this activity to the overall ambient noise background of the ice-free shallow-water Beaufort environment will be quantified with a variety of metrics, in terms of both level (peak-to-peak, rms, sound exposure level), frequency, and time (intervals and fraction of time present). During some years, up to four airgun operations could be detected simultaneously, and a random one-second time sample yielded a 40% chance of containing an airgun signal, but the levels detected are generally within the bounds of natural wind-driven ambient noise levels. This dataset provides useful empirical insight into discussions about the cumulative effects of anthropogenic activity on an environment extensively used by several marine mammal species. [Work sponsored by the Shell Exploration and Production Company.]

5:00–5:45 Panel Discussion

Session 2pBA

Biomedical Acoustics and Signal Processing in Acoustics: Biomedical Applications of Acoustic Standing Waves

Martin Wiklund, Chair
Applied Physics, Royal Institute of Technology, Stockholm 10691, Sweden

Chair's Introduction—1:00

Invited Papers

1:05

2pBA1. Acoustic focusing flow cytometry. Gregory Kaduchak, Gregory R. Goddard, and Michael D. Ward (Molecular Cell Biology Engineering, Life Technologies, 29851 Willow Creek Rd, Eugene, OR 97402, greg.kaduchak@lifetech.com)

Acoustic cytometry is a new technology that replaces or partly replaces hydrodynamic focusing of cells or particles in flow cytometry with forces derived from acoustic radiation pressure. The ability to focus cells into a tight line without relying on hydrodynamic forces allows many possibilities outside the scope of conventional flow cytometry. Dilute samples can be processed quickly. Flow velocities can be varied allowing control of particle delivery parameters such as laser interrogation time and volumetric sample input rates. Recently, Life Technologies unveiled a flow cytometer that directs particles into the laser interrogation region using acoustic radiation pressure. In this talk, the application of acoustic cytometry in flow cytometry systems from fundamental principles to details of its implementation will be presented. Data will be shown for both the operational implementation of the acoustic focusing device as well as demonstrating its ability to perform for complex biological assays.

1:25

2pBA2. Acoustophoretic cell sorting in microfluidic channels. Tom H. Soh and Allen Yang (Mechanical Engineering, UC-Santa Barbara, Santa Barbara, CA 93111, tsoh@engr.ucsb.edu)

In this work, we report the use of ultrasonic acoustophoresis for the label-free separation of viable and non-viable mammalian cells within a microfluidic device. This device exploits the fact that cells that have undergone apoptosis are physically smaller than viable cells, and achieves efficient sorting based on the strong size dependence of acoustic radiation forces in the microchannel. As a model, we have selectively enriched viable MCF-7 breast tumor cells from heterogeneous mixtures of viable and non-viable cells. We found that this mode of separation is gentle on cells, while enabling label-free separation at sample flow-rates of up to 12 mL/hr at a cell density of 10^6 cells/mL. We have extensively characterized the device and report the effects of piezoelectric voltage and sample flow-rate on device performance, and describe how these parameters can be tuned to optimize recovery, purity or throughput.

1:45

2pBA3. Standing surface acoustic wave technology for cell and organism manipulation. Sz-Chin S. Lin and Tony J. Huang (Department of Engineering Science and Mechanics, The Pennsylvania State University, University Park, PA 16802, szchinlin@gmail.com)

Manipulation of particles, cells, and organisms is essential to many fundamental biological studies. For example, the ability to precisely control the physical settings of biological objects allows scientists to investigate the interactions between molecules, cells, or cell and its environment. Acoustic-based particle manipulation techniques possess non-contact and non-invasive natures and have rapidly become key enablers for many emerging lab-on-a-chip biomedical applications. We, BioNEMS lab at Penn State, have developed a series of standing surface acoustic wave (SAW) based acoustic tweezers that can dexterously manipulate a wide range of cellular-scale objects. This technology uses standing SAW induced acoustic radiation forces to trap suspended biological objects, whose physical location/orientation can then be dynamically changed by adjusting the parameters (e.g. frequency and amplitude) of the standing SAW. Simply by tuning the AC electrical signals, our system can perform sophisticated cell patterning, focusing, reorientation, separation, sorting, transportation, and stretching without direct contact. We show that our system can be seamlessly integrated with other on-chip devices and is fully compatible with fluorescence and confocal microscopies. The versatility, simplicity, low power consumption, non-contact and non-invasive natures render our system an excellent platform for a wide range of applications in the biological, chemical, and physical sciences.

2:05

2pBA4. Ultrasound standing wave fields for tissue engineering. Denise C. Hocking (Pharmacology and Physiology, University of Rochester, 601 Elmwood Ave., Box 711, Rochester, NY 14642, denise_hocking@urmc.rochester.edu), Kelley A. Garvin, and Diane Dalecki (Biomedical Engineering, University of Rochester, Rochester, NY)

The spatial organization of cells within native tissues contributes to proper tissue function. Thus, successful engineering of replacement tissues requires methods to control cell location within the engineered tissue. Our studies have focused on utilizing acoustic radiation forces associated with ultrasound standing wave fields (USWF) to rapidly and non-invasively organize cells within 3D collagen

gels. We have shown that USWF-induced alignment of fibroblasts into distinct multicellular planar bands increases cell contractility and enhances cell-mediated extracellular matrix reorganization. Additionally, USWF-induced patterning of endothelial cells accelerates the formation of capillary-like sprouts and supports the maturation of sprouts into lumen-containing, vascular networks throughout the volume of the collagen gel. Our recent studies have investigated the influence of various acoustic parameters on the USWF-induced spatial pattern of endothelial cells. The initial density of the USWF-induced cell bands affected both the rate of formation and the morphology of endothelial cell networks, indicating that different USWF-induced endothelial cell patterns can produce morphologically different vascular networks. Design of USWF, by choice of ultrasound frequency or use of multiple transducer geometries, can create more complex cell patterns within hydrogels. Thus, USWF technologies provide a novel approach to pattern large 3D engineered tissues in vitro.

2:25

2pBA5. Applications in acoustic manipulation of biological cells in micro-devices. Dyan N. Ankrett, Peter Glynne-Jones, and Martyn Hill (Engineering Sciences, University of Southampton, University of Southampton, University Road, Southampton, Hampshire SO17 1BJ, United Kingdom, M.Hill@soton.ac.uk)

Utilizing ultrasonic standing waves in biocompatible micro-fluidic devices, we are able to acoustically manipulate biological cells for a variety of potentially beneficial pharmaceutical, biomedical and environmental applications. Using a device designed to induce sonoporation in the absence of contrast agent micro-bubbles (CA-free sonoporation) we demonstrate both the uptake and efflux of differently sized, membrane impermeable molecules whilst maintaining high cell viability. Crucially we show that the cytotoxic action of several known pharmaceutical agents is significantly increase in porated cells compare with non-porated control cells, suggesting sonoporation-induced facilitated uptake of these agents. We also report on a device that acoustically excites tethered polymer-shelled micro-bubbles to induce micro-streaming around cardiomyocyte membranes in order to mimic myocardial infarction, ischaemia and induction of apoptosis. Initial results from a levitation culture system that continuously perfuses a pellet of ultrasonically suspended chondrocytes are also presented. The pellet geometry allows large numbers of cells to be cultured without developing a necrotic core. Primary chondrocytes and cell lines demonstrated good viability after more than ten days of levitation. Using low power, continuous ultrasonic excitation we have demonstrated significant reductions in biofilm growth in polymer fluid channels potentially increasing the lifespan of in-situ marine sensors.

2:45

2pBA6. Microfluidic air-liquid cavity acoustic transducers for point-of-care diagnostics applications. Abraham Lee and Maulik Patel (Biomedical Engineering, UC Irvine, 3120 Natural Sciences II, Irvine, CA 92697-2715, aplee@uci.edu)

Microfluidic devices with “side channels” that trap air enable acoustic energy coupling and acoustic streaming into the main channel. This basic configuration is versatile and can be designed as a microfluidic pump, mixer, particle trap, cell sorting switch, and sample separation component. These multiple functions are integrated on a microfluidic platform and provides rapid and specific diagnostics of infectious diseases based on the immune response detected in a drop of blood. A second implementation of this concept is to utilize microfluidic produced lipid microbubbles that respond to acoustic energy as ultrasound contrast agents with drug carrying payload and molecular ligand targeting that can detect and treat molecular diseases such as cancer and cardiovascular diseases. This talk will introduce both of these microfluidic acoustic transducer projects in my lab.

3:05–3:20 Break

3:20

2pBA7. Cell lysis using acoustic cavitation bubbles in microfluidics. Tandiono Tandiono, Siew-Wan Ohl (Fluid Dynamics, Institute of High Performance Computing, 1 Fusionopolis Way, Singapore 138632, tandiono@ihpc.a-star.edu.sg), Cara Sze-Hui Chin, Dave Siak-Wei Ow (Bioprocessing Technology Institute, Singapore), and Claus-Dieter Ohl (Physics and Applied Physics, Nanyang Technological University, Singapore)

Analysis of intracellular contents, such as proteins and nucleic acids, in a micro-scale system is gaining its importance in biomedical research. However, an efficient cell lysis needs to be achieved before the analysis can be carried out. The standard lysis methods in microfluidics, for example: by means of chemicals, thermal, or electrical lysis, suffer from the undesirable temperature increase and cross-contamination, which may lead to the denaturation of proteins or interfere with subsequent assays. Here, we present a technique to mechanically lyse microbial cells using acoustically driven cavitation bubbles in a polydimethylsiloxane based microfluidic channel attached on a glass slide. The cavitation bubbles are created by exciting gas-liquid interfaces in the microchannel into nonlinear interface instability with ultrasonic vibrations. The strongly oscillating bubbles create regions with intense mixing and high shear stress, which can deform and rupture the nearby cells. *Escherichia coli* (bacteria) and *Pichia pastoris* (yeast) cells are completely lysed in less than 0.4 seconds and 1.0 second, respectively. The temperature increase of the samples during the ultrasound exposures is less than 3.3 °C. Fluorescence intensity measurements and real-time polymerase chain reaction (qRT-PCR) analysis suggest that the functionality of the harvested protein and genomic DNA is maintained.

3:40

2pBA8. Structures formed by ultrasonic standing waves in active fluids and suspensions. Mauricio Hoyos (PMMH, CNRS, ESPCI, 10 rue Vauquelin, Paris 75005, France, hoyos@pmmh.espci.fr), Angelica Castro, Eric Clément, Annie Rousselet (PMMH, CNRS, Paris, Ile de France, France), Despina Bazou (Steel Lab, Massachusetts General Hospita, Charlestown, MA), Wei Wang, and Thomas Mallouk (Center for Solar Nanomaterials, Penn State University, State College, PA)

The acoustic radiation force concentrates particulate materials at the nodes or antinodes of an ultrasonic standing wave (USW), in function of different physicochemical parameters: size, shape, density or elastic properties. Thus, frequencies from 0.5 to 10 MHz are adapted for manipulating micron-sized particles, cells, bacteria, vesicles, drops, bubbles and even colloidal species. The interaction of different species with the ultrasonic radiation field generates levitation, aggregation or coalescence. In this presentation, new behaviors

will be presented in active fluids: bacteria bath and self-propelled metallic micro-cylinders. Leaving bacteria in culture medium undergoing the USW field show a dynamics inducing complex structures. USW propel, rotate, align and assemble metallic micro-rods (2 μm long and 330 nm diameter) in water as well as in solutions of high ionic strength, generating “self-acoustophoresis”. Finally, new possibilities for controlling aggregation forming 2D and 3D particle and cancer cells structures using pulsed ultrasound will be shown.

4:00

2pBA9. Acoustic trapping with seed-particles for submicron particle enrichment. Björn Hammarström, Simon Ekström, Thomas Laurell, and Johan Nilsson (Measurement Technology and Industrial Electrical Engineering, Lund University, P.O. Box 118, Lund 221 00, Sweden, bjorn.hammarstrom@elmat.lth.se)

Acoustic trapping in disposable borosilicate capillaries utilize ultrasonic forces to capture/retain micro-particles or cells against fluid flow in a microfluidic-channel. A miniaturized ultrasonic transducer is used to locally excite a 4-MHz cross-sectional resonance in the capillary, creating an acoustic field gradient for retention of cells in non-contact mode. Due to competition between fluidic drag from induced acoustic streaming and primary radiation force the smallest particle size addressable with the trapping system is limited. Here, the typical transition occurs at single-micron particle diameters. However, trapping of single- or sub-micron biological species has highly relevant applications such as enrichment or purification of bacteria or viruses. This work investigates the influence of in-trap particle concentration on the trapping, and it is found that elevated concentrations allow capture of submicron particles. By preloading the acoustic trap with micron-sized seed-particles capture of submicron particles even at low concentrations is enabled. Using this technique, we demonstrate single event capture of bacteria as well as capture of 100nm particles. To provide analytical readout for identification/analysis of the trapped particles the acoustic trap is interfaced with a MALDI-MS instrument. Here, the acoustic trapping capillary is operated in aspirate/dispense mode allowing easy and flexible handling of small sample volumes.

4:20

2pBA10. Ultrasonic standing waves for dynamic micro-array cytometry. Martin Wiklund, Athanasia Christakou, Mathias Ohlin, and Björn Önfelt (Applied Physics, Royal Institute of Technology, KTH-Albanova, Stockholm 106 91, Sweden, martin.wiklund@biox.kth.se)

We describe a novel platform for dynamic micro-array cytometry (DMAC), i.e., parallel screening of individual cell-cell interactions based on ultrasonic standing wave aggregation and positioning of cells in a multi-well microplate. Upon ultrasound actuation, clusters containing one or a few cells are quickly formed and retained in a precise location synchronously in each of the 100 wells on the microplate. By combining the acoustic cell handling tool with high-resolution fluorescence microscopy, detailed time-lapse monitoring of individual cell-cell interactions in a highly parallel manner is possible. Of particular interest in our group is to study the long-term interaction between natural killer (NK) cells and different target cells at the level of single cells. In this talk we demonstrate both theoretically and experimentally how to design a microchip capable of trapping and positioning individual cells by ultrasound in a highly parallel manner, and with a spatial accuracy of the order of a cell diameter. We quantify the cell cluster motility with and without retained ultrasound exposure during 17 h, and we report on the viability of cells when exposed to continuous ultrasound for up to three days. Finally, we quantify the heterogeneity of NK cells’ cytotoxicity against cancer cells.

Contributed Papers

4:40

2pBA11. Acoustic radiation force on a sphere in tissue. Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Applied Research Laboratories, University of Texas, Austin, TX 78713-8029, hamilton@mail.utexas.edu)

A theory is presented for the acoustic radiation force on a sphere embedded in a soft elastic medium that possesses a shear modulus μ several orders of magnitude smaller than its bulk modulus. Scattering of both compressional and shear waves is taken into account. There is no restriction on the size of the sphere or, apart from axisymmetry, the form of the incident compressional wave. The analysis employs the Piola-Kirchhoff pseudostress tensor and Lagrangian coordinates. In the linear approximation an analytical solution is obtained for the scattered waves. The nonlinear stress and full radiation force are calculated at the next order of approximation. For a small sphere and $\mu\approx 0$ the classical result for a particle in liquid is recovered. For small but finite shear modulus the radiation force is evaluated for a gas bubble driven at a frequency below resonance. The predicted magnitude of the radiation force on the bubble is found to be less than that in liquid by the factor $[1+(4/3)\mu/\gamma P_0]^{-1}$, where P_0 is the ambient pressure and γ the ratio of specific heats of the gas. Influence of the scattered shear wave in this limit is negligible. [Work supported by NIH DK070618.]

4:55

2pBA12. Development of a computational model to predict cranial resonance shifts due to changes in intracranial pressure. Andrew A. Piacsek and Sami Abdul-Wahid (Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

A possible method for noninvasively monitoring changes in intracranial pressure is to measure changes in skull resonance frequencies. Recent measurements of the vibrational response of a spherical aluminum shell clearly demonstrate that resonance frequencies shift higher as the internal pressure is increased [Piacsek et al, J. Acous. Soc. Am, 131, EL506-510 (2012)]. The frequency shift is approximately linear with the applied pressure, regardless of whether the shell is filled with air or water, and circumferential modes exhibit larger resonance shift than longitudinal modes. A computational model of a fluid-filled thin shell subject to acoustic stimulation was developed using the COMSOL multi-physics software to investigate the role of shell material and geometry in resonance shifts. The model predicts frequency shifts comparable to those observed in the spherical aluminum shell. Preliminary computational results for a spherical shell made of bone-like material, as well as for asymmetric and nonuniform shells, will be presented.

5:10

2pBA13. Numerical analysis of nonlinear standing waves in two-dimensional acoustic resonators. Fangli Ning (School of Mechanical Engineering, Northwestern Polytechnical University, 127 Youyi Xilu, Xi'an, Shaanxi 710072, China, ningfl@nwpu.edu.cn), Xiaofan Li (Department of Applied Mathematics, Illinois Institute of Technology, Chicago, IL), and Juan Wei (School of Communication Engineering, Xidian University, Xi'an, Shaanxi, China)

High amplitude nonlinear standing waves in acoustic resonators have been used in many engineering applications ranging from microfluidic devices for biomedical research, acoustic compression, to acoustic seal. The distribution of physical properties inside two-dimensional resonators is very useful for developing the application of nonlinear standing waves. Most of the previous numerical studies were limited to one-dimensional resonators, and analyzed standing waves with an assumption: the physical variables are assumed as finite sums of basic functions. In the study, the formation of the standing waves in the two-dimensional resonators is computed in the time domain from the initial position at rest without any predefined standing waves. The two-dimensional unsteady compressible Navier-Stokes equations and the state equation for an ideal gas are employed. This study extends the traditional pressure based finite volume SIMPLEC scheme for solving the equations. Initially, the pressure waves predicted in a two-dimensional cylindrical resonator are excellent agreement with the results obtained with pervious numerical methods in particular finite element and finite difference method. Next, we also investigate the velocity waveforms, and find that the sharp velocity spikes appear at the two ends of the resonator. Finally, the distribution of physical properties inside a two-dimensional cylindrical resonator is obtained.

5:25

2pBA14. Compound manipulation of micro-particles using a single device:Ultrasonic trapping, transporting, and rotating. Kun Jia, Keji Yang, Jian Chen, and Jianxin Meng (Department of Mechanical Engineering, Zhejiang University, Zheda Road No. 58, Hangzhou 310027, China, jiakun@zju.edu.cn)

Ultrasonic manipulation is widely used as a noncontact technology and recently small particles rotating on a vibrating substrate has been observed. In this report,a novel methodology which compounds the procedures of ultrasonic trapping, transporting and rotating micro-particles in fluid using a single device is investigated. Irregular micro-particles in a standing wave field experience both acoustic radiation force and torque, which drive the particles to pressure nodes and keep them in a balance posture. A prototype device has also been built according to this theory, which six phase-controlled piezoelectric transducers whose sound beam axes are arranged with an angle of 60 deg in the x-y plane are used to generate ultrasonic standing waves with arbitrary node. The transducers are divided to two groups, so the wave field can be rotated by switching between the two groups. The synthesized sound field is scanned using a needle hydrophone and 200 μm irregular SiO_2 particles are used to perform the compound manipulation ability of our device. The experimental results show good agreement with the theoretical calculation and the possible reason accounting for the small deviations is also discussed. This method may provide more complex and elaborate applications in micro-assembling and cell biomechanics.

TUESDAY AFTERNOON, 23 OCTOBER 2012 SALON 7 ROOSEVELT, 1:30 P.M. TO 3:30 P.M.

Session 2pEA

Engineering Acoustics: General Topics in Engineering Acoustics

Jerry H. Ginsberg, Chair
School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332

Contributed Papers

1:30

2pEA1. Collocation analysis of higher mode coupling in waveguides with discontinuous cross-section. Jerry H. Ginsberg (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332, jerry.ginsberg@comcast.net)

Standard techniques for analyzing transmission and reflection at discontinuities in a waveguide use weighted residuals to match modal series on each side of the junction. Convergence of the model equations generally is slow, which led Homenticovschi and Miles [JASA, 128 (2010) 628-638] to introduce a renormalization technique. Such a formulation does not lend itself well to implementation of a general computational algorithm. Slow convergence of the standard approach a consequence of using weighted residuals, which is a smoothing process, to interrogate a discontinuity. Evidence of this may be found in the analysis of scattering from cylinders having a discontinuous surface impedance [Ginsberg, JASA, 92 (1992) 1721-1728]. That work showed that a weighted residual formulation converges slowly, whereas a proper description of the field, including the singular nature of the velocity at the discontinity, is obtained from a collocation solution that satisfies the surface boundary condition at a discrete set of points. The present work uses standard modal series to represent the signal in each side of the discontinuity. The junction conditions of the waveguide are satisfied at a dscrete number of points. The rate of convergence will be assessed for a variety of schemes for selecting the collocation points, and the implementation of the formulation as part of a computational algorithm of networks will be discussed.

1:45

2pEA2. Dynamic impedance matching for active noise control in a cylindrical waveguide. Dong Joo Lee, Jae-Wan Lee (School of Mechanical Engineering, Yonsei Univ., Seoul, Republic of Korea), Jae Jin Jeon, Young Soo Seo (Agency for Defense Development, Changwon, Gyeongsangnam-do, Republic of Korea), and Won-Suk Ohm (School of Mechanical Engineering, Yonsei Univ., Engineering Building A586, Yonsei-ro, Seodaemun-gu, Seoul 120-749, Republic of Korea, ohm@yonsei.ac.kr)

In this paper, the feasibility of dynamic impedance matching for noise reduction in a cylindrical waveguide is demonstrated. An active acoustic coating, inserted parallel to the direction of wave propagation, is assumed to dynamically match the acoustic impedance of the incoming wave. The active coating appears as an acoustic branch containing the same fluid, and therefore part of the incoming wave can be diverted to and dissipated in the coating. The performance of the active coating is evaluated using a finite element analysis, where the coating is modeled as a dynamic impedance matching boundary. Simulations reveal that significant reductions in the downstream acoustic pressure can be achieved. Unlike the conventional active techniques that employ phase cancellation, dynamic impedance matching has a number of advantages such as a relatively low power requirement.

2:00

2pEA3. Underwater sound transmission through water tunnel barriers. Matthew J. VanOverloop and David R. Dowling (Mechanical Engineering, University of Michigan, University of Michigan, Ann Arbor, MI 48109, mjvanoverloop@gmail.com)

In hydrodynamic test facilities with flowing water, receiving hydrophones are commonly placed behind a solid barrier to reduce flow noise from buffeting and turbulence. However, acoustic waves from the sound sources under study, such as cavitation bubbles, may propagate through the barrier as compression or shear waves, and the presence of the second wave type distorts signals recorded by the hydrophones. Such distortion depends on the structural characteristics of the barrier and the source-receiver geometry, and it may lead to sound-source detection and localization errors. This presentation describes results from a combined experimental and computational effort to understand the sound transmission characteristics of plastic and metal barriers typically used in water-tunnel testing. The experiments were conducted in a 1.0-meter-deep and 1.07-m-diameter cylindrical water tank using a single sound projector, a receiving array of 16 hydrophones, and impulsive (100 micro-second) signals having center frequencies from 30 kHz to 100 kHz. Computations intended to mimic the experiments are completed with the wave number integration software package OASES. Together the computations and experiments allow the most important barrier parameters to be identified. Dependencies of the received signal on the barrier parameters are presented. [Supported by NAVSEA through the Naval Engineering Education Center.]

2:15

2pEA4. Estimation of parameters of a head-related transfer function customization model. Kenneth J. Faller and Kathleen Hoang (Computer Engineering, California State University, Fullerton, 800 N. State College Blvd., Fullerton, CA 92831, jfaller@fullerton.edu)

Head-Related Transfer Functions (HRTFs) are special digital filters used to create the effect of three-dimensional (3D) virtual sound source placement over headphones. The two most common methods of obtaining HRTFs are to either individually measure the HRTFs on specialized equipment (individualized HRTFs) or to create a set of generic HRTFs by measuring them on a mannequin with average anatomical features (generic HRTFs). Individualized HRTFs required specialized equipment that is not readily available to the general public. Additionally, it is known that HRTFs are heavily dependent on our anatomical features. As a result, generic HRTFs produce significant localization errors. A multi-linear model is now available which uses simple anthropometric measurements of the intended user's anatomy to generate customized HRTFs. These customized HRTFs can be generated without specialized equipment and have improved spatialization over generic HRTFs. However, the anthropometric measurements, which are used as parameters for the customization model, are currently collected manually. In the present work, image processing techniques are used to automatically estimate a portion of the anthropometric measurements of the human pinnae. Analysis of the estimation technique's performance will also be conducted.

2:30

2pEA5. On the secondary path of headset active noise cancellation systems. Buye Xu, Jinjun Xiao (Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com), Thomas E. Miller, Erik Wiederholtz, Daniel M. Warren (Knowles Electronics, Itasca, IL), and Tao Zhang (Starkey Hearing Technologies, Eden Prairie, MN)

Active noise cancellation (ANC) technologies have been successfully applied in headsets to reduce ambient noise in the ear canal. The performance of such a technology depends on the characteristics of the secondary path (SP). In this paper, two different systems are studied: one system utilizes a moving-coil speaker as the secondary source and forms a closed cavity over the ear, while the other one uses a balanced-armature receiver inside the ear canal as the secondary source. The differences in the SP responses of these two systems are investigated both experimentally and theoretically. The implications for designing an effective ANC solution will be discussed based on numerical simulations.

2:45

2pEA6. Free-field reciprocity calibration of low-frequency ultrasonic transducers. M. R. Serbyn (Physics, Morgan State University, 1700 E. Cold Spring Ln., Baltimore, MD 21251, m_rserbyn@msn.com)

Techniques developed for the calibration of electroacoustic transducers provide many practical applications of the principles studied in advanced undergraduate and introductory graduate courses. This communication describes an implementation of the standard method of reciprocity calibration as presented in the popular text, Fundamentals of Acoustics by Kinsler et al. The measurements were performed on inexpensive off-the-shelf transducers designed to operate near 25 kHz and 40 kHz. The anechoic environment consisted of a 2-m3 box lined with absorbing material whose absorption coefficient was measured by the pulse-echo technique. The air density and the speed of sound were calculated using formulas available in the literature. The resulting values of transducer sensitivity (calibration factor) compared well, within 5 -10 %, with the values specified by the vendor. No accuracy statement was available for the transducers under test, not only because of their price, but mainly because no primary calibration services are available in the low ultrasonic range of frequencies, in contrast to the higher, MHz, range. A recent dissertation by N. Bouaoua, discovered in the course of this research, could fill this void should its procedures be adopted by a standards laboratory.

3:00

2pEA7. Acoustic supercoupling through a density-near-zero metamaterial channel. Romain Fleury, Caleb F. Sieck (Department of Electrical and Computer Engineering, University of Texas, Austin, TX), Michael R. Haberman (Applied Research Laboratories, University of Texas, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu), and Andrea Alù (Department of Electrical and Computer Engineering, University of Texas, Austin, TX)

Originally demonstrated with electromagnetic waves, supercoupling describes the extraordinary matched transmission, energy squeezing, and anomalous quasistatic tunneling through narrow channels. This behavior is the result of impedance matching achieved when the effective properties within the channel approach zero. For electromagnetic waves, supercoupling is observed when the electric permittivity in the channel approaches zero. These channels are accordingly known as epsilon-near-zero (ENZ) channels. This work shows that analogous behavior exists in the acoustic domain when the effective density is nearly zero, which can be achieved by tailoring the structure of the channel. Such channels are therefore known as density-near-zero (DNZ) metamaterial channels. Unlike tunneling based on Fabry-Perot resonances, DNZ transmission is independent of channel length and geometry and yields a uniform field along the entire length of the channel. Transmission-line theory is used to describe this peculiar phenomenon and finite element simulations are presented to confirm the unusual transmission properties of the metamaterial channel. It is further shown that acoustic waves may provide a unique possibility of squeezing acoustic energy through arbitrarily small channels in three dimensions, overcoming limitations that arise in the electromagnetic case.

3:15

2pEA8. Acoustic condition monitoring of wind turbines: Tip faults. Daniel J. Comboni and Bruno Fazenda (Acoustics Research Centre, University of Salford, Salford, Greater Manchester, United Kingdom, danielcomboni@gmail.com)

As a significant and growing source of the world's energy, wind turbine reliability is becoming a major concern. At least two fault detection techniques for condition monitoring of wind turbine blades have been reported in early literature, i.e. acoustic emissions and optical strain sensors. These require off-site measurement. The work presented here offers an alternative non-contact fault detection method based on the noise emission from the turbine during operation. An investigation has been carried out on a micro wind turbine under laboratory conditions. 4 severity levels for a fault have been planted in the form of added weight at the tip of one blade to simulate inhomogeneous debris or ice build up. Acoustic data is obtained at a single microphone placed in front of the rotor. Two prediction methods have been developed and tested on real data: one based on a single feature - rotational

frequency spectral magnitude; and another based on a fuzzy logic interference using two inputs - spectral peak and rotational peak variation with time. Results show that the single spectral peak feature can be used to determine fault severity in ranges. The implementation of the fuzzy logic using a further input feature is shown to significantly improve the detection accuracy.

TUESDAY AFTERNOON, 23 OCTOBER 2012 ANDY KIRK A/B, 1:00 P.M. TO 2:00 P.M.

Session 2pED

Education in Acoustics: Take 5's

Jack Dostal, Chair
Physics, Wake Forest University, Winston-Salem, NC 27109

For a Take-Five session no abstract is required. We invite you to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome. You sign up at the door for a five-minute slot before the session starts. If you have more than one demo, sign up for two non-consecutive slots.

TUESDAY AFTERNOON, 23 OCTOBER 2012 ANDY KIRK A/B, 2:00 P.M. TO 4:50 P.M.

Session 2pMU

Musical Acoustics and Education in Acoustics: Teaching Musical Acoustics to Non-Science Majors

Jack Dostal, Chair
Physics, Wake Forest University, Winston-Salem, NC 27109

Invited Papers

2:00

2pMU1. An introduction to the physics of the clarinet for musicians and other non-science majors. Wilfried Kausel (Inst. of Music Acoustics, Univ. of Music and performing Arts, Anton-von-Webern-Platz 1, Vienna 1030, Austria, kausel@mdw.ac.at)

Teaching physics or more generally science related matters to non-science majors requires illustrative comparisons, fascinating application examples, live experiments, multi-medial presentation and simplified representations which can be understood without much background knowledge but which still remain relevant for the taught subject. But first of all, such lectures do require a very dedicated and alert lecturer who is able to captivate his audience by his outstanding presence. Following (some of) these principles a short generally understandable lecture on the acoustics of the clarinet will be given."

2:20

2pMU2. The Physics of Music course as an introduction to science. Gordon Ramsey (Physics, Loyola University Chicago, 6460 N Kenmore, Chicago, IL 60626, gprspinphys@yahoo.com)

Our Physics of Music course is an integration of physics and music. We start with the mathematical structure of music, including scales, harmonies and chords. We discuss musical styles and how they differ. After an introduction of physics concepts, including waves, resonances, forces, energy and fluid flow, the physical structure of instruments in the various groups are studied. Connection is made of the instruments and how they reproduce the mathematical nature of music. Finally, venue acoustics are investigated. The course integrates different styles of learning by integrating different learning modes. The classes include lecture/demonstration, discussion, in-class laboratories and a final individual project encompassing many course elements. The constant connection between the physics and the music, along with varied learning techniques, including hands-on experience, provides a motivating approach for non-science majors to experience science in a familiar context.

2:40

2pMU3. Fostering research in a general education acoustics course. Peter L. Hoekje (Physics and Astronomy, Baldwin Wallace University, 275 Eastland Rd., Berea, OH 44017, phoekje@bw.edu)

The general education science course often is the last science course an undergraduate takes. Liberal arts core requirements may include the goal that students come to understand or appreciate the nature of science and its impact on society. Incorporating research experience may be desirable, but has its challenges in a course for the non-major. A basic toolkit for student inquiry gives the

capabilities for sound editing and analysis, filtering, sound level monitoring, sound and music synthesis, vowel and speech analysis, musical instrument design, and room acoustics analysis. These tools are available as low cost or free software. A course design compromise must be reached between delivery of content and support of student inquiry. However, a class of forty pursuing independent research projects in acoustics makes a rewarding experience for both instructor and students!

3:00

2pMU4. Acoustics at Berklee College of Music and elsewhere. Anthony K. Hoover (McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, CA 91362, thoover@mchinc.com)

Students in the acoustics class at Berklee College of Music were non-science majors, but the course has done well, spanning over 50 semesters, and a number of those students have continued into graduate school and careers in acoustics. This paper will focus on the design and goals of the class, and will also discuss related experiences of teaching non-scientists in seminars, short courses, courses at other colleges, and the ASA-sponsored Continuing Education short course for architects.

Contributed Papers

3:20

2pMU5. Beats, ratios, and commas: Teaching tuning and temperament to music students. Stewart Carter (Music, Wake Forest University, 1833 Faculty Dr., Winston-Salem, NC 27106, carter@wfu.edu) and Jack Dostal (Physics, Wake Forest University, Winston-Salem, NC)

Undergraduate music students have many preconceived notions about tuning and temperament. For most of them, a piano tuned in equal temperament is perfectly "in tune" and A4 = 440 Hz is the only pitch standard that has ever existed. We find these naïve conceptions about tuning and temperament to be common among our introductory Physics of Music students–music majors, students pursuing majors in the sciences, and other non-science majors. In this paper we describe how we address some of these issues. We teach the mathematical underpinnings of equal, Pythagorean, just, and meantone temperaments, as well as their musical advantages and disadvantages, through the use of live demonstrations and recorded examples. We also describe our future plans to use recorded examples and available local resources—including reproductions of Baroque woodwinds tuned to A4 = 415 Hz and a 1799 organ at A4 = 409 Hz—to give students a broader understanding of historical and modern pitch standards.

3:35

2pMU6. Teaching and learning musical acoustics in a "music technology" program. Robert C. Maher (Electrical & Computer Engineering, Montana State University, 610 Cobleigh Hall, PO Box 173780, Bozeman, MT 59717-3780, rob.maher@montana.edu)

Helping math-averse students to learn musical acoustics can be challenging. Many universities offer a degree program entitled "Music Technology." While the details of Music Tech programs differ, the general curricular focus includes music theory, electronic and computer music, audio recording and mixing, multimedia production, and computer applications in music composition. Music Tech programs also typically require a "Science of Sound" course that is intended to introduce the physical and mathematical concepts of musical sound and the basic principles of room acoustics. But unlike traditional acoustics students with a physics or engineering background, many students in the music tech programs are unconventional learners who do not tend to have much background in mathematics and the traditional science topics that would be the customary prerequisites for a formal acoustics class. This situation provides an interesting challenge–and a great opportunity–for the instructor to help students learn about acoustics and audio topics while side-stepping the students' disdain for mathematical formulae. This paper presents one Music Tech lesson example: teaching musical instrument acoustics using a lumped source-filter-coupling model. The learning outcomes are sufficient for the students to understand and implement useful empirical models and simulations.

3:50

2pMU7. Measuring brass instruments: A "Physics of Music" lab exercise. Randy Worland (Physics, University of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Among the various aspects of brass instruments studied in a Physics of Music class are physical dimensions such as tube length and bore profile. Although textbooks list values for instrument lengths and describe the significance of the cylindrical, conical, and flared sections of tubing, these parameters are not visually obvious in the coiled instrument. A laboratory exercise is described in which students make simple length and diameter measurements of real brass instruments. Plots of tube diameter vs. distance are produced that effectively display what each instrument would look like if straightened out. Students also determine the air paths through valves (which are surprisingly difficult to visualize), and measure the added tubing associated with each valve. Mouthpieces are examined and measured in order to demonstrate the complex set of shapes involved. These hands-on measurements force students to look closely at the component pieces, air paths, and construction details of the instruments. Pedagogically, the use of real instruments in the lab engages students and helps them make connections between the physics of ideal tubes and the design of real brass instruments. Student results and feedback will be described.

4:05

2pMU8. Introducing the concept of resonance to non-science majors in a musical acoustics class. Andrew Morrison (Joliet Junior College, 1215 Houbolt Rd, Natural Science Department, Joliet, IL 60431, amorrison@jjc.edu)

Understanding the concept of resonance is one of the most important goals for a student in a musical acoustics course to achieve. In our musical acoustics course students examine the standing wave behavior of simple one-dimensional systems such as stretched strings and pipes with open ends or one closed end. The concept is integral to the discussion of how nearly all musical instrument work, but it is also a difficult concept for many students to understand. In our class, we discuss how looking at the boundary conditions of the systems we examine in class can be applied to predict what the resonant frequencies of the system are.

4:20

2pMU9. An integrated lecture-laboratory approach to teaching musical acoustics. Andrew A. Piacsek (Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

At Central Washington University, "Physics of Musical Sound" (PHYS103) is an introductory level course that satisfies a General Education Breadth requirement in the lab-based science category. Enrollment is capped at 40 students, of which 10-15 are typically music education majors. From 1998 to 2010, this five-credit course was taught in a traditional lecture format: four 50-minute lectures and a two-hour lab each week. Starting in 2011, the course was reformatted into three two-hour periods. The objective was to incorporate inquiry-based learning strategies adopted from physics education research. The two-hour meeting time provides flexibility to structure the class with a combination of lecture, tutorial, group problem solving, and experiment. Computational resources, including sound analysis software and online interactive simulations, are heavily utilized. Example activities will be described and lessons learned from the transition to the new format will be discussed.

4:35

2pMU10. Physics of music for musicians, architects, and science students: Designing a multi-level platform for an undergraduate course.
Robert G. Hall (Music, Laurentian University, 935 Ramsey Lake Rd., Department of Music, Sudbury, ON P3E2C6, Canada, rhall@laurentian.ca)

This presentation will describe the development of a new course in the Physics of Music at Laurentian University in Sudbury, Ontario. While the creation of the course was spurred on by the opening of a new School of Architecture in 2013 and the desire to provide an elective containing acoustics for that program, the project also has the added challenges of: developing a course that is available to music students (many of whom have little or no science background), creating a course that is an attractive elective for students across campus, initiating a course that meets the elective requirements for engineering students, and building a course that contains sufficient physics so that it may be cross-listed as a physics course. The presentation will outline the specifics of each of the required parameters and give specifics as to the multi-level project choices within the course that have been developed to allow the different student constituencies to satisfy their pedagogical needs. Also addressed will be the requisite teaching methods used to allow on-line participation, while still encouraging and requiring attendance at the weekly lectures.

TUESDAY AFTERNOON, 23 OCTOBER 2012 — TRIANON C/D, 1:30 P.M. TO 5:00 P.M.

Session 2pNS

Noise and Architectural Acoustics: Soundscape Workshop

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Sq., Vernon, CT 06066

Jian Kang, Cochair
School of Architecture, Univ. of Sheffield, Sheffield S10 2TN, United Kingdom

Chair's Introduction—1:30

Invited Papers

1:35

2pNS1. Introduction to workshop goals.
Bennett M. Brooks (Brooks Acoustics Corporation, 30 Lafayette Square - Suite 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com), Brigitte Schulte-Fortkamp (Technical University Berlin, Berlin, Berlin, Germany), and Jian Kang (School of Architecture, University of Sheffield, Sheffield, South Yorkshire, United Kingdom)

The goal of this workshop is to work toward methodology standardization for the advancement of the developing field of soundscape measurement, analysis and design. The workshop will focus on the terminology lexicon used for soundscape subject interviews. Interviews of local experts, residents and other users and inhabitants of the sonic environment can yield insights into both personal reactions and universal observations. The specific terminology used in this process may significantly affect the outcomes. As the success of a new research or development project can depend on the lessons learned from previous projects, the standardization of interview techniques becomes increasingly important. Workshop participants are invited to develop a standardized lexicon of descriptors for field use in interview questionnaires. The lexicon development will be based on the results of earlier ASA soundscape workshops, and the concurrent activities in ISO and COST Working Groups. An introductory keynote address will review the topics, objectives, and procedures for the day's discussion. The participants will then break out into smaller subgroups to review key issues and to assess the available lexicon terms. The entire group will reassemble to report, assess, and prioritize the proposed actions.

2:00–3:00 Break Out Session

3:00–3:30 Break

3:30–4:30 Break Out Session

4:30–5:00 Panel Discussion

Session 2pPA

Physical Acoustics: Waves in Heterogeneous Solids II

Joseph A. Turner, Cochair
Dept. of Mechanical and Materials Engineering, Univ. of Nebraska-Lincoln, Lincoln, NE 68588-0526

Goutam Ghoshal, Cochair
Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801

Invited Papers

1:00

2pPA1. Ultrasonic scattering for inverse characterization of complex microstructures. Stanislav I. Rokhlin (Department of Material Science and Engineering, The Ohio State University, Edison Joining Technology Center, 1248 Arthur E Adams Drive, Columbus, OH 43220, rokhlin.2@osu.edu)

This paper overviews resent theoretical and experimental results for inverse characterization of duplex polycrystalline microstructures from measurements of ultrasonic attenuation and backscattering. The duplex microstructures are formed by elongated regions of clustering crystallites with preferred orientations and are typical for titanium alloys. New insights into the dependences of the backscattering and attenuation on frequency and averaged ellipsoidal grain radii are obtained. In particular the dominant effect of the averaged ellipsoidal radius in the direction of wave propagation, instead of the ellipsoidal cross-section, is shown. Also the opposite effect on attenuation and backscattering of grain size in the direction of wave propagation is demonstrated by theory and experiment. Both attenuation and backscattering depend on effective elastic properties of clusters, their size and orientation and the measurement system parameters. It is shown that for material property inversion it is advantageous to decouple elastic and geometrical properties by a special series of measurements and data inversion: first determine grain geometry and size from relative directional and/or attenuation-to-backscattering ratio measurements; this is followed by absolute attenuation and backscattering measurements to obtain the crystallite orientation distribution function in the clusters.

1:20

2pPA2. Ambient noise seismic monitoring of the continuous deformation of the Earth. Michel Campillo (ISTerre, Université Joseph Fourier, BP 53, Grenoble 38041, France, Michel.Campillo@ujf-grenoble.fr)

The analogy between the Green function and the long-range correlation of the seismic ambient noise leads to new developments in imaging and monitoring. In the last years this approach allows for high-resolution surface wave tomography, and more recently its potential for body wave imaging was demonstrated. In the same time, the reconstruction of the scattered part of the Green function was analyzed and its possible use to detect slight variations in the medium was confirmed. Ambient noise monitoring allows for a continuous measure of seismic velocity changes related with the tectonic process affecting the lithosphere. I present some examples showing that a change at depth can be monitor with seismic noise records and that those changes are related to the deformation. We analyze in detail the transient deformation on a subduction zone and we discuss the relations between change of seismic velocity, slow slip events and tremors.

1:40

2pPA3. Multiple scattering in the spirit of Leslie Foldy. Paul A. Martin (Applied Mathematics and Statistics, Colorado School of Mines, Dept of Applied Math and Statistics, Colorado School of Mines, Golden, CO 80401-1887, pamartin@mines.edu)

Foldy's name is best known to wave theorists because of his 1945 paper on multiple scattering. This came out of wartime work on sound propagation through bubbly liquids. The paper itself contains a deterministic theory for scattering by a finite number of small objects, and a probabilistic theory for wave propagation through random arrangements of many small scatterers in which a certain closure assumption is invoked. Assumptions of this kind can be said to be unreasonably effective. We describe some of this early work and then review more recent work on a variety of multiple scattering problems.

2:00

2pPA4. Acoustic scattering from weakly coupled porous media. Max Denis, Kavitha Chandra, and Charles Thompson (University of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, max_f_denis@hotmail.com)

In this work, wave propagation and multiple acoustic scattering from porous media are examined. For a highly porous media, the inertial coupling between solid and fluid phases can be small. In such a case the dilatational and acoustic motions will be weakly coupled. The controlling factor is the ratio of effective inertial mass density of the porous media and the fluid mass density. Using Biot's analysis as the starting point, the dilatational and acoustic motions, as well as the scattered pressure that ensues are evaluated. The Kirchhoff-Helmholtz integral equation is used to this end. The integral equations are evaluated using a series expansion in terms of derived acoustic contrast properties. Using the local Padé approximant procedure detailed in [The Journal of the Acoustical Society of America, 128(5) EL274-278 (2010)]. Of particular interest, are high contrast media and wavelengths comparable to the average scatterer size.

2:15

2pPA5. Characterization of polycrystals by ultrasonic attenuation-to-backscattering ratio measurements. Jia Li and Stanislav I. Rokhlin (Department of Material Science and Engineering, The Ohio State University, Edison Joining Technology Center, 1248 Arthur E Adams Drive, Columbus, OH 43220, rokhlin.2@osu.edu)

Scattering of ultrasonic waves in polycrystalline materials depends on the relative misorientation of the crystallites, their elastic properties and the crystallite size and morphology. Those parameters affect both ultrasonic attenuation and backscattering. In structural material alloys the elastic properties of crystallites and microtextures are often unknown, thus complicating model-based microstructure size determination from ultrasonic measurements. It has been shown by analysis and simulation that the measured attenuation-to-backscattering ratio is independent of the unknown elastic material characteristics and thus may be used advantageously for inverse determination of mictostructural characterization of polycrystalline materials. The method is based on recently developed models for ultrasonic attenuation and backscattering in polycrystals with nonequiaxed grains of general ellipsoidal shape. Using the model approximation an effective size parameter is defined and obtained from the backscattering-to-attenuation ratio measurements. By this method the microstructure size from one side of the sample can be characterized. The method is demonstrated on experimental data for Ti alloy samples with duplex microstructure/microtexture formed by hexagonal alpha crystallites. Volumetric ultrasonic results are compared with direct surface microtexture measurements by orientation image microscopy.

2:30

2pPA6. Ultrasonic characterization of polymeric composites containing auxetic inclusions. Michael R. Haberman, Daniel T. Hook, Timothy D. Klatt (Applied Research Laboratories, University of Texas, 10000 Burnet Rd, Austin, TX 78758, haberman@arlut.utexas.edu), Trishian A. M. Hewage, Andrew Alderson, Kim L. Alderson (Institute for Materials Research and Innovation, University of Bolton, Bolton, 5AB, United Kingdom), and Fabrizio L. Scarpa (Bristol Laboratory of Advanced Dynamics Engineering (BLADE), Department of Aerospace Engineering, University of Bristol, Bristol, 1TR, United Kingdom)

Composite materials are often used as damping treatments or structural materials to mitigate the effects of unwanted vibration and sound. Recent work on materials displaying a negative Poisson's ratio, known as auxetic materials, indicate that composite materials consisting of a lossy matrix containing auxetic inclusions may lead to improved vibro-acoustic absorption capacity compared to composites containing positive Poisson's ratio inclusions. This work presents ultrasonic measurements of an epoxy matrix material (Epon E828/D400) containing volume fractions of α-cristobalite inclusions ranging from 5% to 25% by volume. The effective frequency dependent speed of sound and attenuation coefficient of each sample is measured from 1 to 10 MHz using ultrasonic immersion techniques. Ultrasonic test results are compared with Dynamic Mechanical Thermal Analysis and modal damping measurements of stiffness and loss behavior. This material is based upon work supported by the U. S. Army Research Office under grant number W911NF-11-1-0032.

2:45

2pPA7. Estimation the mean correlation length of metals by using mode-converted diffuse ultrasonic backscatter model. Ping Hu and Joseph A. Turner (Department of Mechanical and Materials Engineering, University of Nebraska-Lincoln, Lincoln, NE 68588, ping.hu@huskers.unl.edu)

Diffuse ultrasonic backscatter measurements can explain the phenomenon of the scattering at interfaces in heterogeneous materials. The theoretical models of diffuse ultrasonic backscatter for longitudinal (L-L) and shear (T-T) wave scattering within polycrystalline materials have recently been developed. A mean correlation length of metals is successfully calculated from the L-L model at normal incidence in a pulse-echo inspection. Above the first critical angle, mode-conversion scattering occurs when the longitudinal waves are converted to shear waves (L-T) at material grain boundaries. With a similar formalism, a mode-conversion scattering (L-T) model is presented to describe this process. The model is then to fit the experimental response for a pitch-catch transducer configuration and the correlation length is extracted by modifying the beam function. The mean correlation length from the L-T model is in agreement with both the L-L model and the results obtained from optical micrographs. This presentation outlines the theoretical framework and the method to extract the mean correlation length. Mode-converted backscatter removes the influence of the front-wall reflection and may lead to improvements in microstructural characterization and material property evaluation. [Research supported by FRA.]

3:00–3:15 Break

3:15

2pPA8. Detection of inclusions embedded in a polycrystalline medium: A diagrammatic approach. Lucas W. Koester and Joseph A. Turner (Mechanical and Materials Engineering, University of Nebraska-Lincoln, Lincoln, NE 68588, lucaswkoester@huskers.unl.edu)

Diffuse ultrasonic backscatter techniques are used to evaluate and to quantify polycrystalline microstructure in structural materials including steel, aluminum, and concrete. Modeling of the backscatter received from grain scattering is important for both microstructural characterization and flaw detection. Here, the problem of an inclusion embedded in a polycrystalline background is examined using a diagrammatic approach. Expressions are derived that for the attenuation and diffuse backscatter using a Green's function approach that incorporates Feynman diagrams for simplicity in the analysis. The resulting equations are solved in the single-scattering limit that couples geometric probabilities and correlation functions such that the impact of the inclusion can be quantified. The effects of frequency, average grain size, polycrystalline properties, inclusion properties and inclusion size are considered. Further work related to inspection with focused beams or immersion type ultrasonic inspections is also presented. This work is anticipated to impact the studies of model-assisted probability and inclusion content quantification in ultrasonic non-destructive evaluation.

3:30

2pPA9. Diffuse ultrasonic backscatter measurements for monitoring stress in rail. Devin T. Walentine, Christopher M. Kube, Joseph A. Turner (Mechanical and Materials Engineering, University of Nebraska, Lincoln, NE 68588, d.t.walentine@gmail.com), and Mahmood Fateh (Office of Research and Development, Federal Railroad Administration, Washington DC, Nebraska)

Recent research in polycrystalline materials, both theoretically and experimentally, has demonstrated a dependence of diffuse ultrasonic backscatter (DUB) on applied stress. This dependence has been used to develop a measurement device for monitoring longitudinal stress in continuous

welded rail (CWR). However when moving this research from the laboratory setting to field applications many additional challenges are encountered with respect to the experimental method. In this presentation, results from initial field experiments are presented. This work is performed on a switch track from a railroad mainline to a short line railroad siding. Comparison between DUB measurements and actual stress in the rail is accomplished through the use of four stress modules (based on standard strain gauge technology) that were installed at four different locations along test site. In these field tests, DUB experiments using two orthogonal shear waves are investigated. Procedures to mitigate errors in our experiments and techniques to refine the measurement for more accurate results are discussed. These field experiments highlight the utility of this approach with respect to practical and financial considerations for determining induced stresses in CWR. [Research supported by FRA.]

3:45

2pPA10. Influence of stresses on grain size and attenuation parameters in ultrasonic scattering models. Christopher M. Kube and Joseph A. Turner (University of Nebraska-Lincoln, Lincoln, NE 68588, ckube@huskers.unl.edu)

Recent findings have shown the dependence of material stresses on the strength of ultrasonic backscatter. The stress dependence arrives when considering the covariance of effective elastic moduli of a medium with a randomly oriented grain structure. In some instances, the change in magnitude of the scattered response due to stress is significant and can be utilized as a technique for NDT stress monitoring. Conversely, ignoring the stress sate could result in significant error when attempting to extract micrsotructural parameters using ultrasonic NDE techniques. This presentation touches on the error generated in estimating the average grain size when using ultrasonic scattering models which include stress dependent backscatter coefficients. Results are given for three scattering modes; longitudinal to longitudinal, shear to shear, and (mode converted) longitudinal to shear under a variety of loading configurations. Furthermore, the stress dependence in the covariance influences common multiple echo attenuation measurements when grain scattering is present. Theoretical attenuation values of the different modes are given for various stress-states. Understanding how stress influences these parameters can potentially improve existing ultrasonic NDT techniques. [Research supported by the Federal Railroad Administration.]

4:00

2pPA11. Vibrations of composite bimorph cantilever with multidomain structures. David Sedorook (Department of Physics and Astronomy, University of Mississippi, University, MS), Andrew Nadtochiy (Department of Physics, Kyiv Shevchenko University, Kyiv, Ukraine), and Igor Ostrovskii (Dept. of Physics and NCPA, Univ of Mississippi, Lewis Hall, Rm. 108, University, MS 38677, iostrov@phy.olemiss.edu)

The vibrating piezoelectric cantilevers are important objects in contemporary physical acoustics and their applications including the hot topic of robotic flying insects, MEMS, and various actuators. In this work, we study different cantilevers consisting of the piezoelectric layers and a thin steel layer. The three-layer bimorph composite cantilever includes a central steel sheet covered with two oppositely polarized coatings made of a piezoelectric ceramic. We compare the amplitudes of tip vibrations in the bimorph composite structures with the simpler cases of a single piezoelectric layer with one to three ferroelectric domains sitting on a steel substrate as well as without the steel support. Theoretical calculations of the amplitude of acoustical vibrations of the actuator tip are performed with the help of a finite element method developed for piezoelectric media. The corresponding codes were verified and adopted by the experimental measurements from the 12-mm long and 0.5-mm thick PZT-cantilever vibrating on its fundamental mode at frequency 1.12 kHz. The vibration amplitude of a cantilever with 2 or 3 periodically poled domains is the highest at the 2nd or 3rd resonance frequency, respectively.

4:15

2pPA12. Spatial variation of ultrasonic attenuation coefficient in freshly excised human calvarium. Armando Garcia, Joseph A. Turner (Mechanical and Materials Engineering, University of Nebraska-Lincoln, Lincoln, NE 68588-0526, aagn17@gmail.com), and Robert S. Salzar (Center for Applied Biomechanics, University of Virginia, Charlottesville, VA)

The propagation of ultrasound through human calvarium poses a great challenge for transcranial diagnosis and treatment of several medical conditions. Moreover, better understanding of how sound waves are attenuated in the human calvarium is gaining relevance with the recent awareness of the problem of blast wave induced traumatic brain injury. In the present study, the spatial variability of ultrasonic properties was evaluated for relevant frequencies of 0.5, 1, and 2.25 MHz. A total of eighteen specimens from four donors were tested using a through-transmission configuration. With the aid of a two interface model, the ultrasonic attenuation coefficient was determined from the total energy loss at various locations on the specimens. Mean volumetric densities at various locations on the samples were determined from computed tomography images. The results show good correlation between attenuation and volumetric density, particularly for the higher frequencies. In addition, the spatial variability of the attenuation, within a single person and with respect to different people, was found to be much larger than expected. These results are anticipated to have a major impact on transcranial biomedical research. [Support of the Army Research Office for this work is gratefully acknowledged.]

4:30

2pPA13. Exact image theory for a three layer medium. Ambika Bhatta, Charles Thompson, Kavitha Chandra (Electrical and Computer Engineering, University of Massachusetts, 1 University Ave, Lowell, MA 01854, ambika_bhatta@student.uml.edu), and Vineet Mehta (MIT Lincoln Laboratory, Lexington, MA)

In this work an exact formulation and solution for the image source amplitude in a three layer medium is given. To do so a novel generalization of the Sommerfeld integral is employed. Of particular interest in this paper is the time-domain solution for the impulse response between source and observation points in the middle layer. It is shown that global boundary conditions can be accommodated for multiple reflections from media of having infinite extent. A branch integral formulation of inverse Laplace transform of integral powers of the Fresnel reflection coefficient is given for this case. It is shown that only odd-powers of the branch point argument contribute thereby reducing computational effort required to numerically evaluate the impulse response.

4:45

2pPA14. A sonar experiment to study sound propagation through flames. Mustafa Z. Abbasi, Preston S. Wilson (Mech. Eng. Dept. and Applied Res. Labs., University of Texas, Austin, TX 78712, mustafa_abbasi@utexas.edu), Ofodike A. Ezekoye, and Joelle I. Suits (Mech. Eng. Dept., Univ. of Texas at Austin, Austin, TX)

Disorientation is a major cause of firefighter death and injury. When a firefighter is trapped in a structure, there is a small window of time for rescue teams to locate a downed firefighter. We propose an acoustic navigation system based on sonar concepts to augment existing techniques (e.g. thermal imaging cameras). As a first step toward this system, a better understanding of acoustic wave propagation through the fire environment is needed. Therefore, a sonar experiment was developed to measure the distance through a flame. Since information in the literature suggests that transmission loss through fire increases with frequency, a parametric array was used to maintain narrow source directivity while remaining in the low frequency/low attenuation regime. Results of the sonar experiment will be presented for various source functions and flame conditions.

Session 2pSA

Structural Acoustics and Vibration: Guided Waves for Nondestructive Evaluation and Structural Health Monitoring II

Henrique Reis, Cochair
Industrial and Enterprise Systems Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801

Michael J. Lowe, Cochair
Mechanical Engineering, Imperial College London, London SW7 2AZ, United Kingdom

Invited Papers

1:00

2pSA1. Integral structural health monitoring of helicopter tail booms by guided acoustic waves. Wolfgang Grill (Institute of Experimental Physics II, University of Leipzig, Linnéstr. 5, Leipzig, Germany 04312, Germany, grill@physik.uni-leipzig.de)

The general principles of structural health monitoring by ultrasonic waves with high resolution monitoring of the time dependence of the transport properties are presented. Reported are essential aspects of the theoretical background together with the results of modeling relating to the developed monitoring scheme and the experimental findings. The presented applications of the developed integral structural health monitoring concentrate on examples involving helicopter tails booms. These include conventional aluminum based bolted frame and stringer constructions as well as fiber compound structural components with foam inlays, in both cases for helicopter tail booms. The presentation includes experimental demonstrations on samples suitable for transport.

1:25

2pSA2. Hybrid microcontinuum field approach for intrinsic damage state quantification. Sourav Banerjee (Mechanical Engineering, University of South Carolina, 300 Main Street, Columbia, SC 29208, banerjes@cec.sc.edu)

The primary objective of this presentation is to demonstrate a systematic framework for the early stage diagnostics of materials under operation. Defying the conventional route of 'bottoms-up' multi-scale modeling approach for material diagnostics, a hybrid 'top-down' approach is presented. Structure specific diagnostics and prognostics have become extremely important due to variances in the real life performances of aerospace materials. Structures, structural components and engine components will need individual attention in the near future. Individual attention means that a parallel database must be maintained for each component for on-line digital certification. The 'digital-twin' software will have access to structure's detailed fleet and maintenance records, nondestructive test results, real-time sensory data etc. to certify the materials on-line, but it will need some 'quantifiable' parameters. Thus a parameter to quantify the incubation of damage at meso-scale is identified. The intrinsic length scale dependent 'parameter called 'damage entropy' closely resembles the material state due to fatigue, extreme environments, operational hazards or spatio-temporal variability. The proposed quantification process involves physics and statistics based predictive models. In this novel approach 'micromorphic' description of material is considered in spatio-temporal non-local sense and the nonlocal features are extracted from the real time signals for damage state quantification.

1:50

2pSA3. Mesh-free distributed point source method for modeling guided wave propagation in a viscous fluid layer trapped between two solid plates. Yuji Wada (Precision and Intelligence Laboratory, Tokyo Institute of Technology, Yokohama, Nagatsuta, Midori-ku, Japan), Tribikram Kundu, and Kentaro Nakamura (Civil Engineering and Engineering Mechanics, University of Arizona, 1209 E. 2nd Street, Bldg # 72, Tucson, AZ 85721, tkundu@email.arizona.edu)

Distributed Point Source Method (DPSM) is extended to model wave propagation in viscous fluids. Appropriate estimation on attenuation and boundary layer due to fluid viscosity is necessary for the ultrasonic devices that utilize acoustic streaming or ultrasonic levitation. Since the boundary layer is often much thinner than the wavelength, numerical simulations based on the finite element method suffer from large computational cost because very fine mesh is needed to trace the layer. DPSM can efficiently model this problem. The computational cost for modeling the viscous fluid with DPSM is reduced close to the cost of non-viscous fluid analysis. In this paper, equations for DPSM modeling in viscous fluids are derived by decomposing the linearized viscous fluid equations into two components - dilatational and rotational. By considering complex P-wave and S-wave numbers, the sound fields in viscous fluids can be calculated using the same calculation routines used for waves in solids. To verify the calculation precision, a comparison between approximated theory and DPSM generated results for a fundamental ultrasonic field problem is performed. The particle velocity profile parallel to the surface in a viscous fluid between two vibrating plates is calculated. Theoretical results agree well with the DPSM generated results.

2:15

2pSA4. Active structural health monitoring during sub-orbital space flight. Andrei N. Zagrai, William Reiser, Brandon Runnels, Chris White, Abraham Light-Marquez, Stephen Marinsek, Andrew Murray (Mechanical Engineering, New Mexico Institute of Mining and Technology, 801 Leroy Pl., Socorro, NM 87801, azagrai@nmt.edu), Stuart Stuart Taylor (The Engineering Institute, Los Alamos National Laboratory, Socorro, NM), Gyuhae Park, and Charles Farrar (The Engineering Institute, Los Alamos National Laboratory, Los Alamos, NM)

Increasing involvement of commercial enterprises in space activities is among leading forces behind a renewed interest in structural diagnostic methodologies promising potential for improving safety, operability and cost effectiveness of launch vehicles and spaceships. It is envisioned that unobtrusive, real time structural health monitoring (SHM) systems may assist in space vehicle's prelaunch qualification, orbital operation, safe disintegration during reentry or recertification for a next flight. SHM experiment utilizing piezoelectric wafer active sensors in conjunction with electro-mechanical impedance measurements has been develop to explore feasibility of active structural health monitoring during suborbital space flight. Details of experiment are discussed and some results obtained in real time for all segments of vehicle's trajectory are presented. Experimental data collected during suborbital space flight has shown feasibility of SHM in the challenging environment, utility of thin wafer piezoelectric sensors as active elements of spacecraft SHM system, and potential of the electro-mechanical impedance method for real time structural integrity assessment of the payload.

2:40

2pSA5. Modeling and signal processing for guided wave structural health monitoring. Anthony Croxford and Paul D. Wilcox (University of Bristol, Queens Building, Bristol, Avon BS8 1TR, United Kingdom, a.j.croxford@bristol.ac.uk)

Guided wave approaches are a key technique for the implementation of structural health monitoring with a sparse sensor array, offering good sensitivity over a reasonable range. There are however several difficulties, particularly how the system can be modelled to predict performance and how acquired data can be used most effectively post capture. This paper looks at both these issues, starting with a study of the modelling of the scattering of guided waves from features. Models of both the scattering from structural features and predicted damage types can then be used to determine the sensitivity of any network and the process of their derivation and use is described here. In order to maximise this sensitivity it is important to use the data in the most efficient way possible. This is accomplished through the use of statistical models of the changes in the signal and fitting maximum likelihood estimators to these. In choosing a good model of the signal a statistical detector can be designed to optimise the performance for a given use case, be that detection or localisation.

3:05–3:20 Break

Contributed Papers

3:20

2pSA6. Elastic wave propagation in coated pipelines. Ray Kirby, Zahari Zlatev (Mechanical Engineering, Brunel University, Uxbridge, Middlesex UB8 3PH, United Kingdom, ray.kirby@brunel.ac.uk), and Peter Mudge (NDT Technology Group, TWI Ltd., Cambridge, United Kingdom)

Guided elastic waves are commonly used in the non-destructive evaluation of oil and gas pipelines. In order to protect pipes from environmental damage a viscoelastic coating such as bitumen is often applied. Viscoelastic coatings do, however, attenuate travelling waves and it is not uncommon in a coated pipe for the signal reflected from a defect to be attenuated to the extent that it is no longer discernable above the background noise. Thus, viscoelastic coatings significantly impact on the effectiveness of non-destructive testing in pipelines, both in the ability to resolve defects but also in the length of pipeline that may be tested. In order to obtain a better understanding of guided wave propagation in coated pipes, a numerical model is presented here that seeks to analyse the propagation of torsional and longitudinal modes in a coated pipe with simple axisymmetric defects. Reflection coefficient predictions are compared with experimental data for a pipe coated with bitumen and good agreement is observed between the two, although only after first undertaking a curve fitting exercise to identity appropriate values for the phase speed and attenuation of bulk torsional waves in the viscoelastic material.

3:35

2pSA7. Characterization of porous materials using ultrasonic slow wave. Lin Lin (Engineering, University of Southern Maine, 37 college Ave., 131 John Mitchell center, Gorham, ME 04038, llin@usm.maine.edu), Michael Peterson (Mechanical Engineering, University of Maine, Orono, ME), and Alan Greenberg (Mechanical Engineering, University of Colorado at Boulder, Boulder, CO)

Porous materials are critical engineering components in many process industries. Although porous materials have been successfully utilized in many areas, characterization of porous structures is still a significant problem limiting the applications of porous materials, especially when the application involves the change of porous structure. Ultrasonic techniques have been reported for successful applications on material characterization, including porous materials. This research utilized an acoustic technique for permeability measurement by measuring the critical wave number of Biot's slow longitudinal wave. The slow longitudinal wave can serve as an important tool for investigating the ability of fluids to penetrate into porous materials. Since the slow longitudinal wave is associated with out-of-phase movement of the pore fluid relative to the matrix structure, it is very sensitive to the permeability of the porous formation. In Biot's theory the slow wave is highly attenuated below a critical frequency. The critical wave number can be directly related to the permeability of porous materials. The measurement of the critical wave number provides a unique method to obtain the permeability measurement, which can be apply to structure monitoring, quality control, etc.

3:50

2pSA8. Guided wave approach for inline photovoltaic module component inspection. Rico Meier (Module Reliability, Fraunhofer CSP, Walter-Huelse-Strasse 1, Halle, Saxony-Anhalt 06120, Germany, rico.meier@csp.fraunhofer.de)

Reliability of photovoltaic modules is one key factor for being financially attractive for customers all over the world. The further reduction in manufacturing costs lead to increased demands on module components and their materials to maintain acceptable mechanical yields and module reliability. Thus fast, economic and preferably non-destructive component characterization and manufacturing process control methods come more and more into focus. In the present work lamb wave based approaches for material characterization of plate-like photovoltaic module components were evaluated with respect to their precision and inline capabilities. The second-order elastic constants (Young's modulus and Poisson's ratio) were determined by automated numeric fitting of the Rayleigh-Lamb dispersion model on experimental data. Furthermore, the influence of mechanical stress on the ultrasound transport properties was investigated. Therefore the components

were systematically loaded by mechanical and thermal stress. The resulting changes in ultrasound transport were correlated with results from mechanical testing. Lamb wave approaches turned out to be well suited methods for inline material characterization of plate-like module components. The elastic constants can be determined with high accuracy. The usage of plate-like structures in photovoltaic modules makes lamb waves an important tool for inline and non-destructive material characterization.

4:05

2pSA9. Determination of leaky Lamb wave modal attenuation coefficient on a solid plate. Wan-Gu Kim and Suk Wang Yoon (Dept. of Physics, SungKyunKwan University, 300 Chunchun-dong, Jangan-ku, Suwon, Suwon 440-746, Republic of Korea, swyoon@skku.ac.kr)

Leaky Lamb wave is practically important in an immersion technique of nondestructive evaluation for a solid plate. It is necessary to determine its modal attenuation coefficient in order to evaluate plate defects in industry and to diagnose osteoporosis in medical applications. In this study we present a method to determine the modal attenuation coefficient of leaky Lamb wave using two-dimensional Fourier filtering. A complex leaky Lamb wave signal from a solid plate can be decomposed into several modal signals on frequency-space domain through two-dimensional Fourier filtering. It makes possible to experimentally determine the modal attenuation coefficient along an aluminum plate in water. It is well explained by theoretical estimation with the dispersion relation of leaky Lamb wave.

4:20

2pSA10. Simultaneous thickness, velocity, density, and attenuation measurement of a thin layer by time-resolved acoustic microscopy. Jian Chen, Xiaolong Bai, Keji Yang, Bingfeng Ju, and Jianxing Meng (The State Key Laboratory of Fluid Power Transmission and Control, Zhejiang University, Yuquan Campus, Hangzhou, Zhejiang Province 310027, China, yi03072004@stu.xjtu.edu.cn)

An ultrasonic method for simultaneous determination of thickness, velocity, density and attenuation of thin layer using a time-resolved acoustic microscopy is proposed. Reflection from the thin layer is represented as a function of three dimensionless parameters which are determined from the experimentally normal incidence component of the two dimensional reflection spectrum $R(\theta,\omega)$ derived based on the inversion of V(z,t) technique with time-resolved acoustic microscopy. The thickness of the thin layer is derived from the signals received from the layer itself considering the geometrical relations when the lens focused on the sample's surfaces. The simultaneous determination of thickness, velocity, density and attenuation of thin stainless steel plate by using a point-focusing transducer with nominal frequency of around 50MHz were carried out. The determined material properties are comparable, in which the thickness, velocity and density can be measured with a percentage biases less than 5% and the attenuation is close to its real value. The present preliminary work shows the high efficiency, viability and capability of the new non-destructive technique in simultaneously characterize basic mechanical and geometrical properties of thin layers.

4:35

2pSA11. Simultaneous measurement of local longitudinal and transverse wave velocities, attenuation, density, and thickness of a layer by using point-focus ultrasonic spectroscopy. Jian Chen, Xiaolong Bai, Bingfeng Ju, and Jianxing Meng (The State Key Laboratory of Fluid Power Transmission and Control, Zhejiang University, Yuquan Campus, Hangzhou, Zhejiang Province 310027, China, yi03072004@stu.xjtu.edu.cn)

This paper presented an ultrasonic technique for simultaneous determination of the complete set of acoustical and geometrical properties of a layer embedded between two known materials using point-focus ultrasonic spectroscopy, which provides a high lateral resolution. The theoretical model of the two-dimensional spectrum Rt (θ, ω) of the layer is calculated as a function of six parameters of longitudinal and transverse velocities cl, ct, attenuation αl, αt, density ρ and thickness h, which fully determined the layer properties. The experimental spectrum Re (θ, ω) can be measured by V(z, t) technique. A two-step algorithm is presented to decompose the searching process of parameters from one six-dimensional to two three dimensional spaces. The sensitivity of the two-dimensional spectrum to individual properties and its stability against experimental noise are studied. The full set properties of a 250 μm thick stainless steel plate immersed in water are determined. The proposed technique used a point focus transducer, which makes the set-up simple and reliable. It allows measurement of the local properties of the layer and enables precision material characterization.

TUESDAY AFTERNOON, 23 OCTOBER 2012 — TRUMAN A/B, 1:00 P.M. TO 5:00 P.M.

Session 2pSC

Speech Communication: Speech Perception I: Vowels and Consonants (Poster Session)

Yuwen Lai, Chair

Foreign Languages and Literatures, National Chiao-Tung University, Taiwan, Hsinchu 30010, Taiwan

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

2pSC1. Context matters: Analyzing the necessity of context-dependent speech perception in complex models. Keith S. Apfelbaum and Bob McMurray (Psychology, University of Iowa, Iowa City, IA 52242, keith-apfelbaum@uiowa.edu)

Speech perception faces variability in the realization of phonemes from vocal tract differences, coarticulation, and speaking rate. While the mechanisms accommodating this variability have long been debated, recent computational work suggests that listeners succeed by processing input relative to expectations about how context affects acoustics. McMurray and Jongman (2011) measured 24 cues for fricatives in 2880 tokens. When effects of talker voice and vowel context were removed with a relative encoding scheme, simple cue-integration/prototype models identified the fricatives at near-human levels of performance. These models were less successful with raw cue values, suggesting some form of compensation is required. However, contingent encoding may not be needed in all categorization architectures. Exemplar models account for context by holistically comparing incoming tokens to others in memory, and connectionist back-propagation models encode stimulus-specific information through their

hidden units for better performance. We conducted several simulations using these classes of models. After training the models to identify fricatives from the McMurray and Jongman corpus, we assessed model performance on held-out tokens for which human categorization data was available. Both classes of models more closely matched human performance when trained on context-adjusted information than raw cues. This offers firmer support for context-dependent speech perception.

2pSC2. Voicing in English revisited: Measurement of acoustic features signaling word-medial voicing in trochees. Joseph C. Toscano (Beckman Institute for Advanced Science and Technology, University of Illinois at Urbana-Champaign, 405 N Mathews Ave, Urbana, IL 61801, jtoscano@illinois.edu) and Bob McMurray (Dept. of Psychology and Dept. of Communication Sciences & Disorders, University of Iowa, Iowa City, IA)

A great deal of work in speech has argued that invariant acoustic cues do not exist, leading many researchers to conclude that listeners use specialized representations, such as talkers' inferred gestures, instead. Other work has emphasized that many phonological distinctions are signaled by multiple cues; Lisker (1986, Language and Speech, 29, 3-11), for example, lists 16 cues to voicing. Yet, few studies have measured the reliability of multiple cues and asked whether combining them may provide a solution to the lack of invariance. Here, we present measurements of 12 potential cues to the voiced/voiceless distinction in stops (including many of those reported by Lisker, 1986) and use a statistical modeling approach to determine which ones distinguish the two categories and how reliable those cues are. We recorded two-syllable non-words (/sVCVs/) with one of six consonants (/b,p,d,t,g,k/) and one of four vowels (/a,æ,i,u/). We found that talkers used multiple cues, but that cues varied in their usefulness. In addition, a classifier trained on the cues was able to accurately identify voicing categories. We argue that by harnessing information from multiple cues, listeners can overcome ambiguity in individual cues in specific utterances, allowing them to recognize speech across talkers and phonological contexts.

2pSC3. Linguistic and social effects on perceptions of voice onset time in Korean stops. Robert Podesva, Annette D'Onofrio, Eric Acton, Sam Bowman, Jeremy Calder, Hsin-Chang Chen, Benjamin Lokshin, and Janneke Van Hofwegen (Linguistics, Stanford University, 450 Serra Mall, Building 460, Stanford, CA 94305, annetted@stanford.edu)

This paper investigates effects of linguistic and social factors on phoneme categorization of Seoul Korean stops. In an investigation of VOT in aspirated versus lenis stops of Korean, Oh (2011) reports VOT length in aspirated stops to be conditioned both linguistically and socially: bilabial stops exhibit shorter VOT than velars, following /a/ conditions shorter VOT than /i/, and female speakers exhibit shorter VOT than males. 10 native speakers of Seoul Korean (5 men, 5 women) were recorded producing bilabial and velar stops in the frame /CVn/. Recordings were manipulated to create a 10-step continuum of VOT length for each speaker, from 25ms to 115ms. 30 native speakers of Seoul Korean listened to each of these manipulated stimuli for every speaker and categorized them as containing either aspirated or lenis stops. Listeners were more likely to categorize a given VOT as aspirated when it occurred in a bilabial stop as opposed to a velar stop, when it preceded /a/ as opposed to /i/, and when it was produced by a female as opposed to a male. Results indicate that speakers exhibit knowledge of production patterns when categorizing phonemes, drawing upon both linguistic and social information.

2pSC4. Speakers of tonal and non-tonal Korean dialects use different cue weightings in the perception of the three-way laryngeal stop contrast. Hyunjung Lee, Allard Jongman, and Stephen Politzer-Ahles (Linguistics, University of Kansas, 1541 Lilac Ln., Lawrence, KS 66044, hyunjung@ku.edu)

The current study investigates if and how listeners from tonal and non-tonal varieties of Korean (Kyungsang and Seoul) use different cue weighting in their perception of the three-way distinction among Korean voiceless stops (fortis, lenis, and aspirated). Forty-two Korean listeners (21 each from Seoul and Kyungsang) were tested with stimuli in which VOT and F0 were systematically manipulated. Analyses of the perceptual identification functions show that VOT and F0 cues trade off each other for the perception of the three stops. However, the trading relationship differs between the two dialects. Logistic regression analysis reported interactions among VOT, F0 and Dialect factors for perceiving one stop over the others, indicating the two dialects use the perceptual cues differently. Specifically, while Seoul listeners rely primarily on F0 for making lenis responses and on VOT and F0 for aspirated responses, F0 plays a less important role in modulating both lenis and aspirated responses for Kyungsang speakers. A similar pattern has been observed in production [Lee and Jongman 2012, JIPA 42 (2)]. The results will be discussed in terms of the close link between perception and production across the two different dialects.

2pSC5. The waiting is the hardest part: How asynchronous acoustic cues are integrated for fricative voicing and place of articulation. Marcus E. Galle and Bob McMurray (Psychology, University of Iowa, Iowa City, IA 52240, marcus-galle@gmail.com)

A fundamental issue in speech perception is the fact that information is spread over time. This raises the question of how listeners integrate acoustic information in real-time. One possibility is that cues are utilized as soon as they arrive to partially activate lexical candidates. Alternatively, input could be a buffered, until sufficient information is available to make a decision. Between these extremes, listeners may vary depending on the usefulness of a given cue, and whether it directly cues a phonetic contrast, or serves as context (e.g., talker identity) for interpreting other cues. We examined this in word-final and initial fricative contrasts using the visual world paradigm. Participants selected a visual referent of an auditory word (ship), and we used the likelihood of fixating lexical competitors (sip/ship) at each point in time to determine when various factors affected higher level decision making. Several studies contrasted the order and utility of the cues (strong vs. weak in different positions), and direct cues vs. information like talker used for compensation. In general, listeners appear to use acoustic information immediately as it becomes available in real-time to lexical candidates.

2pSC6. The contribution of high-frequency information to fine-grained speech perception in cochlear-implant-simulated speech. Ashley Farris-Trimble and Bob McMurray (Dept. of Psychology, University of Iowa, Iowa City, IA 52242, ashley-trimble@uiowa.edu)

It has been observed that the contribution of information in the higher frequency ranges (>7000 Hz) to speech perception is negligible. This assumption has influenced cochlear implant processing strategies, which typically filter out high-frequency information. However, even if speech recognition is robust without high-frequency information, that information may nevertheless contribute to the actual processing of speech. We asked 15 normal-hearing participants to categorize words along 8-step b/p (e.g., beach to peach) and s/sh (e.g., self to shelf) continua in normal and CI-simulated speech. The CI simulations used a bandpass filter with 8 bands between 200 and 7000 Hz. The envelope for each band was extracted and used to amplify broadband noise within the band's frequency range. Results showed that participants' boundaries shifted significantly for both continua; the endpoints of each continuum were categorized consistently, but ambiguous tokens were more often categorized as sh and b when presented in CI-simulated speech. That is, eliminating high-frequency information during simulation also eliminated information necessary to the perception of sounds with high-frequency cues. These results suggest that while the loss of high-frequency information may not affect CI users' ability to recognize speech, it may compromise their fine-grained discrimination.

2pSC7. Perceptual cues in Korean fricatives. Goun Lee and Allard Jongman (Linguistics, University of Kansas, 1541 Lilac Lane, Lawrence, KS 66045-3129, cconni@ku.edu)

The current study explores the production and perception of two Korean fricatives - fortis [s'] and non-fortis (plain) [s]. Production data from 10 speakers were examined to investigate the acoustic cues that distinguish the two fricatives in different vowel contexts (high vowel /i/ vs. low vowel /a/). Measures included rise time, intensity, center of gravity (COG), F0, H1-H2, and CPP, as well as frication, aspiration, and subsequent vowel duration. COG and vowel duration consistently distinguished the two fricatives; additional cues varied across vowel contexts. For the /i/ context, intensity and F0 differed across the fricatives; for the /a/ context, rise time, H1-H2 and

CPP differed across the fricatives. Four perceptual identification experiments were conducted with sixty native Korean listeners. Experiment 1 established that listeners can distinguish the two fricatives in intact natural syllables. In Experiments 2 and 3, listeners heard only the excised consonantal or vocalic segment. For the /a/ context, fricative identification was successful based on both consonantal and vocalic segments. In the /i/ context, fricative identification exceeded chance level only for the consonantal segment. In Experiment 4, cross-spliced stimuli revealed that speakers are more sensitive to vocalic cues than consonantal cues, but only in the /a/ context.

2pSC8. Influence of lexical and acoustic context on phonetic categorization depends on listening situation. Lori L. Holt and Eva Reinisch (Psychology, Carnegie Mellon University, 5000 Forbes Avenue, Pittsburgh, PA 15213, evarei@andrew.cmu.edu)

Phonetic categorization is influenced by multiple sources of contextual information, but little is known about how different sources of information interact. We examined the relative influence of lexical versus acoustic contexts on phonetic categorization of sounds along [s]-[S] continua embedded in word-nonword pairs (e.g., a[S]amed-a[s]amed, ca[s]ino-ca[S]ino). These categorization targets were preceded by sequences of 12 nonspeech tones with mean frequencies a standard deviation above or below the spectral means of the endpoint fricatives. Listeners' [s]-[S] categorization was influenced by lexical information, exhibiting a Ganong effect with categorization shifted toward responses consistent with words, and also by acoustic context. The effect of the tone sequence was spectrally contrastive; there were more [S] responses (low spectral mean) following higher-frequency tones and more [s] responses (high spectral mean) following lower-frequency tones. In addition, the influence of acoustic relative to lexical context was modulated by listening environment. When the informational load of the lexical context was low (four word-nonword continua) acoustic context exerted a relatively greater influence than when the informational load of the lexical context was high (forty word-nonword continua). Multiple sources of context interact to influence phonetic categorization and the relative influence of different information sources is flexibly modulated by listening environment.

2pSC9. "Talker normalization" effects elicited with no change in talker. A. Davi Vitela, Andrew J. Lotto, and Brad H. Story (Speech, Language, and Hearing Sciences, University of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, adv1@email.arizona.edu)

For more than fifty years, it has been known that listeners' perception of a target speech sound may shift as a result of a change in speaker of a preceding carrier phrase. The predominant theories explaining this phenomenon suggest that the listener must be extracting information specific to the speaker - either generating speaker-specific acoustic-phonemic mapping or an estimate of the dimensions of the speaker's vocal tract. To the extent that these processes derive representations that are specific to a speaker, one would not expect shifts in target perception when the apparent speaker remains unchanged. This prediction was explicitly tested by appending targets to different carrier phrases produced by the same speaker. Speaker characteristics were controlled by using an articulatory synthesizer that allows one to specify the source and dimensions of the vocal tract. Significant shifts in target perception were readily obtained, even when the speaker remained the same. These results suggest that these shifts are not tied to talker identity or anatomy but to more general aspects of the context acoustics. [Work supported by NIH-NIDCD.]

2pSC10. Efficient coding of redundancy among formant frequencies in vowels. Christian Stilp (Department of Psychological and Brain Sciences, University of Louisville, University of Louisville, 308 Life Sciences Building, Louisville, KY 40292, christian.stilp@gmail.com) and Keith Kluender (Speech, Language, and Hearing Sciences, Purdue University, West Lafayette, IN)

Stilp and colleagues (*Proc. Natl. Acad. Sci.* [2010]; *JASA* [2011]; *PLoS One* [2012]) demonstrated that auditory perception rapidly and automatically exploits redundancy among acoustic attributes in novel complex sounds. When stimuli exhibited robust covariance between acoustic dimensions (attack/decay, spectral shape), discrimination of sound pairs violating this pattern was initially poorer than that for sound pairs respecting the pattern. While results support efficient coding of statistical structure in the environment, evidence of its contribution to speech perception remains indirect. The present effort examines perceptual organization adhering to statistical regularities in speech sounds. Vowel stimuli (/ɑ/, "ah") were synthesized to reflect natural correlation between formant frequencies across talkers; as vocal tract length decreases (from men to women to children), formant center frequencies increase (here F_1-F_2 varied; others held constant). Listeners discriminated vowel pairs that either obeyed this correlation (16 pairs) or violated it (1 pair) in randomized AXB trials without feedback. Performance replicated results with nonspeech sounds. Vowels that violated natural redundancy between formant frequencies were discriminated poorer than vowels that obeyed this pattern. Results encourage an efficient coding approach to speech perception, as redundancy among stimulus attributes is exploited to facilitate perceptual organization and discrimination. [Supported by NIDCD.]

2pSC11. Timing of perception for all English diphones. Natasha L. Warner (Linguistics, University of Arizona, Box 210028, Dept. Ling, Univ. AZ, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), James McQueen (Max Planck Institute for Psycholinguistics, Nijmegen, Gelderland, Netherlands), Priscilla Z. Liu, Maureen Hoffmann (Linguistics, University of Arizona, Tucson, AZ), and Anne Cutler (Max Planck Institute for Psycholinguistics, Nijmegen, Gelderland, Netherlands)

Information in speech does not unfold discretely over time; perceptual cues are gradient and overlapped. However, this varies greatly across segments and environments: listeners cannot identify the affricate in /ptʃ/ until the frication, but information about the vowel in /li/ begins early. Unlike most prior studies, which have concentrated on subsets of language sounds, this study tests perception of every English segment in every phonetic environment, sampling perceptual identification at six points in time (13,470 stimuli/listener; 20 listeners). Results show that information about consonants after another segment is most localized for affricates (almost entirely in the release), and most gradual for voiced stops. In comparison to stressed vowels, unstressed vowels have less information spreading to neighboring segments and are less well identified. Indeed, many vowels, especially lax ones, are poorly identified even by the end of the following segment. This may partly reflect listeners' familiarity with English vowels' dialectal variability. Diphthongs and diphthongal tense vowels show the most sudden improvement in identification, similar to affricates among the consonants, suggesting that information about segments defined by acoustic change is highly localized. This large dataset provides insights into speech perception and data for probabilistic modeling of spoken word recognition.

2pSC12. Importance of sentence-level and phoneme-level envelope modulations during vowels in interrupted speech. Daniel Fogerty (Communication Sciences and Disorders, University of South Carolina, 1621 Greene St, Columbia, SC 29208, fogerty@sc.edu)

Speech interrupted by noise has been used as a simplified case for listening to speech in the presence of a fluctuating masker. The present study investigated the importance of overall vowel amplitude and intrinsic vowel amplitude modulation to sentence intelligibility. Eight young normal-hearing listeners participated in the experiment. Sentences were processed according to three conditions that replaced vowel segments with noise matched to the long-term average speech spectrum. Vowels were replaced with (1) low-level noise that distorted the overall sentence envelope, (2) segment-level noise that restored the overall syllabic amplitude modulation of the sentence, and (3) segment-modulated noise that further restored faster temporal envelope modulations during the vowel. Results demonstrated incremental benefit with increasing resolution of the temporal envelope. An additional seven listeners participated in a separate experiment that replaced vowels with a vowel masker instead of noise. The vowel masker was modified to have a flattened fundamental frequency at the mean of the replaced segment. The vowel masker either had a standard temporal envelope or was modulated by the envelope of the replaced vowel. Listeners who heard vowel maskers performed more poorly than those who heard noise maskers. No benefit of segment modulation was observed with the vowel masker.

2pSC13. Refining bouba-kiki: Phonetic detail and object dimensionality in sound-shape correspondences. Annette D'Onofrio (Linguistics, Stanford University, 450 Serra Mall, Building 460, Stanford, CA 94305, annetted@stanford.edu)

Speakers cross-linguistically associate non-words that have round vowels, such as /buba/, with round shapes, and non-words without round vowels, such as /kike/, with spiky shapes (e.g. Kohler 1947). While this link has been attributed to cognitive associations between rounded vowel sounds and images of rounded lips, stimuli have conflated vowel roundness with other phonetic features that may also contribute to the correspondence. In this study, 200 listeners matched abstract objects with nonsense words that systematically varied by vowel frontness, consonant place of articulation, and consonant voicing. Listeners are significantly more likely to select a spiky shape over a round shape when given words with voiceless consonants, alveolar consonants, and front vowels; combinations of these features strengthen the effect. These findings are corroborated in the realm of real-world objects. 102 participants were more likely to name a rounded member of a real-world object class when hearing a word containing non-front vowels than when hearing a word containing front vowels. Further, two- versus three-dimensional object roundness influences the strength of this association. This study provides an empirically and phonetically refined perspective on the paradigm, demonstrating the benefit of considering both detailed phonetic correlates and refined object properties in work on sound symbolism.

2pSC14. Tone recognition in continuous Mandarin Chinese. Jiang Wu, Stephen A. Zahorian, and Hongbing Hu (Electrical & Computer Engineering, SUNY-Binghamton, 4400 Vestal Parkway East, Vestal, NY 13850, jiang.wu@binghamton.edu)

Tones are important characteristics of Mandarin Chinese for conveying lexical meaning. Thus tone recognition, either explicit or implicit, is required for automatic recognition of Mandarin. Most literature on machine recognition of tones is based on syllables spoken in isolation or even machine-synthesized voices. This is likely due to the difficulty of recognizing tones from syllables extracted from conversational speech, even for native speakers of Mandarin. In this study, human and machine recognition of tones from continuous speech is evaluated and compared for four conditions: 1, vowel portions of syllables; 2, complete syllables; 3, syllable pairs; 4, groupings of three syllables. The syllables are extracted from the RASC-863 continuous Mandarin Chinese database. The human listeners are all native speakers of Mandarin. The automatic recognition is based on either Hidden Markov Models, or neural networks, and a combination of spectral/temporal, energy, and pitch features. When very little context is used (i.e., vowel segments only) the human and machine performance is comparable. However, as the context interval is increased, the human performance is better than the best machine performance.

2pSC15. Lexical tone in Mandarin spoken word processing. Joan Sereno and Hyunjung Lee (Linguistics, University of Kansas, 1541 Lilac Ln., Lawrence, KS 66044, sereno@ku.edu)

Two priming experiments examined the separate contribution of lexical tone and segmental information in the processing of spoken words in Mandarin Chinese. Experiment 1 contrasted four types of prime-target pairs: tone-and-segment overlap (bo1-bo1), segment-only overlap (bo2-bo1), tone-only overlap (zhua1-bo1), and unrelated (han3-bo1) in an auditory lexical decision task with 48 native Mandarin listeners. Experiment 2 further investigated the minimal segmental overlap needed to trigger priming when tonal information is present. Four prime-target conditions were contrasted: tone-and-segment overlap (pa2-pa2), only onset segment overlap (ping2-pa2), only rime overlap (na2-pa2), and unrelated (kui4-pa2) in an auditory lexical decision task with 68 native Mandarin listeners. Results showed significant priming effects, as compared to the unrelated baseline, when both tonal and segmental information matched. Moreover, when only segmental information overlapped, priming was also observed while no priming for the tone-only overlap condition was found. Interestingly, partial segmental overlap, with matching tonal cues, did not produce priming. In fact, tonal overlap with matching onset segmental information resulted in significant inhibition. These data clearly indicate that tonal similarity only provides facilitation if there is complete segmental overlap. These findings will be discussed in terms of the segmental and suprasegmental information used in word recognition.

2pSC16. The effect of segmental makeup on Mandarin word and tone recognition. Yuwen Lai (Foreign Languages and Literatures, National Chiao-Tung University, Taiwan, 1001 University Road, Hsinchu, Taiwan 30010, Taiwan, yuwen.lai@gmail.com) and Sheng-Hung Wu (Foreign Languages and Literatures, National Chiao-Tung University, Taiwan, Hsinchu)

The present study adopts the gating paradigm to investigate the effect of onset sonorancy, and coda on Mandarin spoken word and tone recognition. Eight tonal quadruplets (all monosyllabic) with different initial segment (obstruent or sonorant), coda composition (no coda, alveolar nasal, or velar nasal) were used as the stimuli. The gates were formed by a 40-ms increment from the beginning of each word. Twenty native listeners from Taiwan were asked to listen to each stimulus, click the corresponding tone number using a mouse, write down the word and then give their confidence rating on the answer. The Isolation point (IP) based on correct word identification (both segment and tone) and the tone isolation point (TIP) were analyzed. The results indicated that tone recognition can be done before the offset of the stimuli. Tone 1 has an earlier IP than Tone 4, followed by Tones 3 and then Tone 2. Sonorant-initial words have a significant earlier IP than obstruent-initial ones. Words without coda have an earlier IP than alveolar nasal, followed by velar nasals. The processing time course for tone and segment and the effect of tonal features on word processing will be discussed.

2pSC17. Do preceding prosodic patterns influence word recognition in Spanish? Tuuli Morrill, Laura C. Dilley, and Jessica Navarro (Communicative Sciences and Disorders, Michigan State University, East Lansing, MI 48824, tmorrill@msu.edu)

Recent work shows lexical recognition in English is influenced by prosodic characteristics of preceding words (e.g., Dilley et al, 2010, *Journal of Memory and Language*). This study investigates whether preceding prosody affects lexical recognition in Spanish, a language with productive lexical stress contrasts (e.g., *saco* 'bag' vs. *sacó* 'he carried out'). In an experiment, each of 30 disyllabic test items was embedded in a carrier phrase; each item was consistent with one of two lexical interpretations that were distinguished by having primary stress on the initial vs. final syllable (e.g., *saco* vs. *sacó*). The prosody of words preceding the test item was resynthesized with one of two fundamental frequency and timing patterns predicted to generate distinct patterns of syllable prominence favoring perception of primary stress on either the initial or final syllable of the test item. The acoustic characteristics of the test item were identical across prosody conditions. If perception of lexical stress depends in part on prosodic characteristics of the context, then prosody preceding test items should influence lexical perception in Spanish. These findings have implications for understanding the perception of lexical stress during word recognition in languages exhibiting productive lexical stress contrasts.

2pSC18. Influence of recent linguistic exposure on the segmentation of an unfamiliar language. Jui Namjoshi (University of Illinois at Urbana-Champaign, Urbana, IL), Annie Tremblay (University of Kansas, 541 Lilac Lane Blake Hall, Room 427, Lawrence, KS 60045-3129, atrembla.illinois@gmail.com), Mirjam Broersma (Max Planck Institute for Psycholinguistics, Nijmegen, n/a, Netherlands), Sahyang Kim (Hongik University, Seoul, n/a, Republic of Korea), and Taehong Cho (Hanyang University, Seoul, n/a, Republic of Korea)

Studies have shown that listeners segmenting unfamiliar languages transfer native-language (L1) segmentation cues. These studies, however, conflated L1 and recent linguistic exposure. The present study investigates the relative influences of L1 and recent linguistic exposure on the use of prosodic cues for segmenting an artificial language (AL). Participants were L1-French listeners, high-proficiency L2-French L1-English listeners, and L1-English listeners without functional knowledge of French. The prosodic cue assessed was F0 rise, which is word-final in French, but in English tends to be word-initial. 30 participants heard a 20-minute AL speech stream with word-final boundaries marked by F0 rise, and decided in a subsequent listening task which of two words (without word-final F0 rise) had been heard in the speech stream. The analyses revealed a marginally significant effect of L1 (all listeners) and, importantly, a significant effect of recent linguistic exposure (L1-French and L2-French listeners): accuracy increased with decreasing time in the US since the listeners' last significant (3+ months) stay in a French-speaking environment. Interestingly, no effect of L2 proficiency was found (L2-French listeners).

2pSC19. Time-course of lexical knowledge use in processing pronunciation variants. Stanislav M. Sajin and Cynthia M. Connine (Psychology, Binghamton University, 4400 Vestal Parkway East, Binghamton, NY 13902, connine@binghamton.edu)

Three experiments examined how listeners process words that are produced with a context conditioned sound change, fricative assimilation. In fricative assimilation, a word final /s/ can change to a /ʃ / when a following word begins with an approximate segment (e.g., dress yacht –> dresh yacht). Fricative assimilation is blocked in other segmental environments such as plosives (e.g., dress boat will not alter the word final /s/). Previous research revealed that phonological variation is acceptable as long as it is licensed by the context in which it is embedded (Gaskell &Marslen-Wilson, 1996). However, there is indication that such licensing effects are largely consigned to particular types of assimilation (e.g., place assimilation) and that other word-final variations (e.g., flap variants), which happen more often in natural speech, have a separate phonological representation that is dependent on the frequency with which the variation occurs (Ranbom, Connine, &Yudman, 2009). The present research addresses how assimilated speech is processed in recognizing spoken words, and, specifically, how different sources of knowledge (lexical and phonological context) are utilized at different times during processing. The results point out that word knowledge is not essential for utilizing the phonological context during recognition, but it permits earlier use of the phonological context in recognizing the pronunciation variant.

2pSC20. The contribution of individual words to the meaning of sentences. Amber M. Veliz and Theodore S. Bell (Psychology, California State University, Los Angeles, P.O. Box 2476, Montebello, CA 90640, amber.veliz2476@yahoo.com)

The studies purpose was assessing the contribution of individual words of a sentence to its overall intelligibility. The dependent variable was number of target words in each sentence correctly identified at each of three possible word positions, initial, middle, and last. The question of interest was how degradation of audibility of the initial word would affect other words by decreasing the semantic context for those later words resulting from the audibility of the first word. Other variables in the design included presentation levels that resulted in overall performance from 50 to 100 percent correct. There was a three way interaction of word position, masker, and presentation level on performance scores. Noise levels were fixed at 60 dB SPL, and speech levels were varied to -3, -6 and -9 signal-to-noise ratio (SNR). Between subject factors in the ANOVA design were SNR (-3, -6, and -9 db), and masker (flat or enhanced).

2pSC21. Cochlea-scaled entropy predicts intelligibility of Mandarin Chinese sentences. Yue Jiang (Speech, Language, and Hearing Sciences, Purdue University, Heavilon Hall, 500 Oval Drive, West Lafayette, IN 47907-2038, jiang23y@gmail.com), Christian E. Stilp (Department of Psychological and Brain Sciences, University of Louisville, Louisville, KY), and Keith R. Kluender (Speech, Language, and Hearing Sciences, Purdue University, West Lafayette, IN)

Cochlea-scaled spectral entropy (CSE; Stilp & Kluender, *PNAS, 107(27)*:12387-12392 [2010]) is a measure of information-bearing change in complex acoustic signals such as speech. CSE robustly predicts English sentence intelligibility even amidst temporal distortion and widely different speaking rates (Stilp, Kiefte, Alexander, & Kluender, *JASA, 128(4)*:2112-2126). However, CSE does not explicitly capture changes in fundamental frequency ($f0$) in any way distinct from that for other aspects of spectral shape (e.g., formant patterns, slope). This property of CSE could limit its ability to predict intelligibility of tone languages that use $f0$ for phonetic and morphological distinctions. The present study assesses the predictive power of CSE for Mandarin Chinese sentence intelligibility. Twenty-five native-Mandarin listeners transcribed Mandarin sentences in which consonant-length (80-ms) and vowel-length (112-ms) segments with either high or low CSE were replaced with speech-shaped noise. CSE reliably predicted listener performance; as greater amounts of CSE were replaced by noise, performance worsened. Results encourage information-theoretic approaches to speech perception, as change and not physical acoustic measures best predict sentence intelligibility across tonal and nontonal languages. [Work supported by NIDCD.]

2pSC22. Effects of noise on adults' word learning. Min Kyung Han, Holly L. Storkel, and Casey Cox (Speech-Language-Hearing, University of Kansas, 3001 Dole Center 1000 Sunnyside Avenue, Lawrence, KS 66045-7555, minhan@ku.edu)

Neighborhood density, the number of similar sounding words, and phonotactic probability, the likelihood of occurrence of a sound sequence, influence word learning by adults in quiet listening conditions. Specifically, Storkel, Armbruster, and Hogan (2006) found that adults learned high density words and low probability words more accurately. This study examined how neighborhood density and phonotactic probability influence adults' word learning in noisy conditions. Fifty-two college students learned nonwords varying in neighborhood density and phonotactic probability at either +8 dB SNR or 0 dB SNR, and learning was measured in a picture naming task. Results showed a significant interaction of density, probability, and noise level. At +8 dB SNR, no effects were significant. At 0 dB SNR, density interacted with probability, showing better learning when density and probability converged. In other words, for low probability words, learning was better when density was also low (i.e., low-low optimal), whereas for high probability words, learning was better when density was also high (i.e., high-high optimal). These results indicate that noise dampens the effects of probability and density in a moderately noisy condition (i.e., +8 dB SNR) and requires a convergence of probability and density in the noisiest condition (i.e., 0 dB SNR).

2pSC23. The influence of a foreign accent on recall of spoken word lists. Marissa Ganeku and Tessa Bent (Department of Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

Talker and speaking rate variability influence spoken word recall (Nygaard et al., 1995; Goldinger et al., 1991; Martin et al., 1989) suggesting that indexical information affects memory processes. The current study investigated how another type of speech variability - the presence of a foreign accent - influenced spoken word recall. Recall of 10 spoken word lists with 10 words each was assessed at two presentation rates. Listeners received words from a native-accented talker, a non-native talker with a strong foreign accent, or a non-native talker with a mild foreign accent. Neighborhood density, neighborhood frequency, and word frequency were balanced across each list; all words were 100% intelligible (Bent, 2010). Recall was analyzed by: (1) accuracy in the correct serial position and (2) accuracy regardless of serial position. The first analysis demonstrated that listeners recalled more words with the slower presentation rate and in the early and late list positions, replicating earlier findings, but there was no talker effect. The second analysis demonstrated that listeners recalled fewer words from the strongly foreign-accented talker than the native talker or the mildly foreign-accented talker. Thus, foreign-accented speech is another source of variability that can influence processing and encoding in memory.

2pSC24. Sentence recognition as a function of the number of talkers in competing multi-talker babble. Kristin Van Engen and Bharath Chandrasekaran (Communication Sciences and Disorders, University of Texas, Austin, TX 78712, kj.vanengen@gmail.com)

Multiple-talker babble is used in studies of speech perception and processing both as a tool for loading perceptual and cognitive tasks and for direct assessment of speech perception in noise. Single interfering talkers and babbles containing few talkers differ dramatically, however, from babbles with high numbers of talkers. While low N-talker babbles provide greater opportunities for "glimpsing" speech targets during spectral and temporal dips in the maskers (i.e., less energetic masking), the linguistic information available to listeners in the maskers themselves also imposes higher-level interference (i.e., informational masking) that may detract from the identification of target speech. In order to assess the relative overall masking effects of N-talker babble for sentence recognition, the current study utilizes a range of maskers varying in N (1-8), as well as speech-shaped noise. Data collected to date show that performance declines as talkers are added to the masker (that is, as energetic masking increases), but that performance in 6-8 talker babble is significantly better than in speech-shaped noise. Individual variability on speech perception in these various maskers is being assessed by comparing performance on the speech-in-noise task to performance on a range of cognitive and psychoacoustic tasks.

Session 2pSPa

Signal Processing in Acoustics: Cognitive Approaches to Acoustic Signal Processing

Grace A. Clark, Cochair
Engineering, Lawrence Livermore National Laboratory, Livermore, CA 94550

R. Lee Culver, Cochair
ARL, Penn State University, State College, PA 16804

Chair's Introduction—1:10

Invited Papers

1:15

2pSPa1. Adaptive modulation and power control for underwater acoustic communications. Andreja Radosevic (Department of Electrical and Computer Engineering, University of California, La Jolla, CA), Rameez Ahmed Rasheed Ahmed (Department of Electrical and Computer Engineering, Northeastern University, 360 Huntington Avenue, Boston, MA 02115, rarameez@ece.neu.edu), Tolga M. Duman (School of Electrical, Computer and Energy Engineering, Arizona State University, Tempe, AZ), John G. Proakis (Department of Electrical and Computer Engineering, University of California, La Jolla, Arizona), and Milica Stojanovic (Department of Electrical and Computer Engineering, Northeastern University, Boston, MA)

In this work we explore the feasibility of a cognitive acoustic communication system that exploits a dynamic closed loop between the transmitter and receiver with the goal of maximizing the information throughput. Specifically, we design a power control mechanism and couple it with an adaptive modulation method based on orthogonal frequency division multiplexing (OFDM). We propose two methods: the first method is adaptive overall power adjustment in which the transmitter modifies the power gain to provide a target SNR at the receiver by exploiting a feedback link in a time-varying channel. The second method adaptively adjusts the modulation level on each individual sub-carrier to achieve a target bit error rate (BER) at the receiver. Crucial to both of these methods is the ability to predict the acoustic channel and the signal-to-noise ratio (SNR) one travel time ahead, which enables adaptive adjustment of the transmit power and modulation level. The performance of the proposed algorithms is demonstrated using an in-air test bed and further verified with real-time at-sea experiments conducted off the coast of Kauai, HI, in June 2011. Experimental results obtained using real-time at-sea experiments show significant savings in power, as well as improvement in the overall information throughput (bit rate) as compared to conventional, non-adaptive modulation with the same power consumption and target BER.

1:40

2pSPa2. Optimizing receiver and source positioning in the ocean: Lessons from nature. Zoi-Heleni Michalopoulou (Department of Mathematical Sciences, New Jersey Institute of Technology, Newark, NJ 07102, michalop@njit.edu)

Extracting information from the ocean using acoustic signals as well as detecting and localizing sources is critical for both defense and environmental applications. Accurate parameter estimation strongly relies on optimal positioning of sources and receivers, whose locations are often the function of the ocean waveguide. Observing marine-mammal vocalizations, one notices that preference is given to transmission and reception at particular depths. A simple sound propagation analysis indicates that use of the specific locations may optimize sound reception. We can make similar choices with man-made systems, calibrating the locations of deployed sound transmission/reception equipment, with optimal detection and estimation in mind. [Work supported by ONR.]

2:05

2pSPa3. Illumination waveform design for non-Gaussian multi-hypothesis target classification in cognitive radar. Ke N. Wang (Electrical and Computer Engineering, Naval Postgraduate School, Livermore, CA), Grace A. Clark (Engineering, Lawrence Livermore National Laboratory, 7000 Ease Ave., L-130, Livermore, CA 94550, clarkga1@comcast.net), and Ric Romero (Electrical and Computer Engineering, Naval Postgraduate School, Monterey, CA)

A cognitive radar (CR) system is one that observes and learns from the environment; then uses a dynamic closed-loop feedback mechanism to adapt the illumination waveform so as to provide system performance improvements over traditional radar systems. Romero et al. recently developed a CR system that performs multiple hypothesis target classification and exploits the spectral sparsity of correlated narrowband target responses to achieve significant performance improvements over traditional radars that use wideband illumination pulses. This CR system was designed for Gaussian target responses. This research generalizes the CR system to deal effectively with arbitrary non-Gaussian distributed target responses via two key contributions: (1) an important statistical expected value operation that is usually evaluated in closed form is evaluated numerically using an ensemble averaging operation, and (2) a powerful new statistical sampling algorithm and a kernel density estimator are applied to draw complex target samples from target distributions specified by both a desired power spectral density and an arbitrary desired probability density function. Simulations using non-Gaussian targets demonstrate very effective algorithm performance.

Contributed Papers

2:30

2pSPa4. Speech enhancement via only mostly blind source separation.
Richard Goldhor, Karen Chenausky (Speech Technology & Applied Research, 54 Middlesex Turnpike, Entrance D, Bedford, MA 01730, rgoldhor@sprynet.com), Suzanne Boyce (Communication Sciences and Disorders, University of Cincinnati, Cincinnati, OH), Keith Gilbert (Speech Technology & Applied Research, Bedford, MA), Sarah M. Hamilton (Communication Sciences and Disorders, University of Cincinnati, Cincinnati, OH), and Joseph Cin (Speech Technology & Applied Research, Bedford, MA)

In environments in which multiple simultaneously-active acoustic sources contribute to sensor responses, Blind Source Separation (BSS) signal processing techniques may be employed to separate (that is, estimate or reconstruct) the signal characteristics of "hidden" sources. Only Mostly Blind Source Separation (OMBSS) involves the estimation of similar sources in important contexts in which non-acoustic information is also available about one or more of the contributing sources. Recently-reported objective source separation performance measures confirm that non-acoustic information can be used effectively to support high-quality separation in situations in which traditional BSS methods perform poorly (e.g., when more sources are active than there are microphones available). Here we present the results of additional perceptual and objective tests showing that OMBSS processing enhances the intelligibility of speech recorded in the presence of multiple simultaneous speech-babble and non-speech maskers.

2:45

2pSPa5. Use of pattern classification algorithms to interpret acoustic echolocation data from a walking-speed robotic sensor platform.
Eric A. Dieckman and Mark Hinders (Dept of Applied Science, College of William and Mary, P.O. Box 8795, Williamsburg, VA 23187, eric.dieckman@gmail.com)

An unresolved issue for autonomous walking-speed robots in unstructured outdoor environments is maintaining situational awareness. One strategy is combining information from different sensors so the robot can function in a variety of conditions and environments. The very low-cost Microsoft Kinect accessory incorporates active infrared and RGB video sensors to provide real-time depth information, as well as a 4-channel microphone array. We validate the Kinect sensors and investigate the combination of infrared (passive and active) and non-linear acoustic echolocation sensors on our mobile robotic sensor platform. By using an acoustic parametric array to generate the audible echolocation signal, a tightly-controlled beam of low-frequency sound can interrogate targets at long distances, while infrared imaging works well in the nearfield and in difficult weather conditions. Sophisticated signal processing techniques are required to combine and interpret the collected data; we present an example using pattern classification on the acoustic echolocation data to differentiate between vehicle types.

3:00

2pSPa6. Real-time active noise control of magnetic resonance imaging acoustic noise.
Chiao-Ying Lin and Jyh-Horng Chen (Electrical Engineering, National Taiwan University, Taipei 104, Taiwan, d95921031@gmail.com)

Magnetic resonance imaging (MRI) is extensively used in clinical and medical researches because MRI is non-invasive and non-radiation. However, using the MRI cause the loud acoustic noise generated by gradient switching. The MRI noise is annoying patients. Therefore, active noise cancellation (ANC) is used to solve this problem in this article. In this article, the main work is to implement real-time system using DSP. The noise generated by each planar imaging (EPI), which is generally used in fMRI researches, is recorded from 3 Tesla MRI for testing ANC system. The EPI signal is regular and predictable, means the signal like a pattern, we using LMS algorithm to training the signal and learning predict the noise spectrum. After the calculated the noise, we output the reverse single to we home-built headset.

TUESDAY AFTERNOON, 23 OCTOBER 2012 LESTER YOUNG A, 3:25 P.M. TO 4:30 P.M.

Session 2pSPb

Signal Processing in Acoustics: Acoustics for Forensics and Identification

Al Yonovitz, Chair
Communicative Sciences and Disorders, University of Montana, Missoula, MT 59812

Chair's Introduction—3:25

Contributed Papers

3:30

2pSPb1. Digital signal processing in forensic acoustics cases.
Al Yonovitz (Communicative Sciences and Disorders, University of Montana, Curry Health Center, Lower Level, Missoula, MT 59812, al.yonovitz@umontana.edu), Herbert Joe (Graduate School of Business, University of Phoenix, Park City, UT), and Joshua Yonovitz (Law, Business and Arts, Charles Darwin University, Tallahassee, Florida)

Forensic acoustics typically considers speaker identification, authenticity analysis, audio enhancement and transcript verification by analyzing and processing audio or speech signals. Other applications in forensic acoustics include different analyses of non-speech audio signals. This presentation will discuss three contentious litigation cases in which various digital processing techniques formed the foundation of scientifically-based conclusions. The first is a products liability case involving the death of a firefighter and his Personal Alert Safety System. When movement of a firefighter ceases, a high-pitched audible alert is emitted. When numbers of these devices were recorded, the task was to determine if the deceased firefighter's device was included within the multiple devices examined. In the second case the order of the discharges of various caliber weapons based on a 9-1-1 call was determined via digital signal processing. The third study utilized processing techniques to differentiate and identify the type of sounds produced in a number

of murder trials. These cases represent the varied challenges in forensic audio and acoustics investigations, and show the difficulty of applying broad theory to practice, the necessity for innovation in the field and the basis for establishing scientific reliability in order to secure admissibility into the courtroom.

3:45

2pSPb2. Randomized sample-level interpolation for audio content manipulation. Samarth Hosakere Shivaswamy, Stephen Roessner, Xiang Zhou, Gang Ren (Dept. of Electrical and Computer Engineering, Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu), Mark Bocko, and Dave Headlam (Dept. of Electrical and Computer Engineering; Dept. of Music Theory, Eastman Sch. of Music, Univ. of Rochester, Rochester, NY)

In this paper we propose a novel audio signal manipulation algorithm based on sample-level interpolations that can generate multiple unique versions of an audio file without creating any perceptual difference. The proposed algorithm enables important applications such as digital rights management and file distribution tracking. The simplest sample-level interpolation method is based on time domain interpolation of fixed-length audio frames. The processing algorithm first segments the audio signal into fixed-length frames. For each frame, we perform a slight time compression or extension using an audio sample interpolation algorithm and then we recombine the manipulated audio samples to form a manipulated version of the original audio files. To enable better security features a randomization program is applied to control the frame-length and manipulation-length using pseudo-random sequences. The result of this algorithm is effectively a form of weak frequency modulation. If the fame size is larger than the compression/extension sample number, these compression/extension manipulations will not produce any audible difference. Various subjective evaluation experiments are conducted to decide the extent of the admissible processing parameters that will not cause noticeable difference in both fixed-length and randomized-length sample manipulation. The authors also provide several implementation examples and a brief summary.

4:00

2pSPb3. Phase discontinuity detection as a means of detecting tampering with speech recordings. Jerome Helffrich and John D. Harrison (Applied Physics, Southwest Research Institute, San Antonio, TX 78228, jhelffrich@swri.org)

In the forensic analysis of speech recordings, it is frequently necessary to validate the authenticity of the recording—for example, to determine if it has been altered for the purpose of changing the meaning of what was spoken. One method for doing this is to exploit an incidental steady tonal content in the recording, and look for sudden, discontinuous changes in the phase of that tone. We report on the development of an algorithm based on this idea, to be used for detecting edits in digital or analog speech recordings. Our development process included generating a corpus from the TIMIT speech database, designing an algorithm to exploit possible phase discontinuities in the recorded signals at edit sites, and testing the algorithm against the corpus with both edited and unedited copies of the sentence material. A statistical analysis of the performance of the algorithm on sentences contaminated by noise and compressed by various digital compression schemes will be given.

4:15

2pSPb4. Speech enhancement by maintaining phase continuity between consecutive analysis frames. Erdal Mehmetcik (ASELSAN, PK.1. Yenimahalle, Ankara 06200, Turkey, emehmetcik@aselsan.com.tr) and Tolga Ciloglu (Electrical and Electronics Engineering Department, Middle East Technical University, Ankara, Turkey)

The aim of speech enhancement algorithms is to increase the quality and intelligibility of noise degraded speech signals. Classical algorithms make modifications in the magnitude spectrum of the noise degraded signal and leave the phase spectrum unchanged. Leaving the phase spectrum unchanged relies on the results of the early listening tests, in which it is concluded that a better phase estimation does not have a significant effect on speech quality. However, a poor phase estimate causes a noticeable distortion in the reconstructed signal. In this work a new phase estimation method (in the voiced segments of speech) is proposed. This method is then incorporated into classical (magnitude based) enhancement algorithms and a new enhancement method is put forward. The performance of the proposed enhancement method is tested with the ITU-T standard PESQ measure and the results are presented.

TUESDAY AFTERNOON, 23 OCTOBER 2012 MARY LOU WILLIAMS A/B, 1:30 P.M. TO 4:30 P.M.

Session 2pUW

Underwater Acoustics: Parabolic Equation Methods and Comparisons

Timothy F. Duda, Chair
Woods Hole Oceanographic Inst., Woods Hole, MA 02543

Contributed Papers

1:30

2pUW1. Exact parabolic equation for the sound field in inhomogeneous ocean. Nikolai Maltsev (Frontier Semiconductor, 2127 Ringwood Avenue, San Jose, CA 95131, admin@asymptotus.com)

A system of equations $\partial P/\partial x = i\omega\rho u, \partial u/\partial x = (i\omega\rho)^{-1}(-\triangle_{yz} + \nabla_{yz}\rho, \nabla_{yz}) - (\omega/c)^2)P$ where $P(\mathbf{r}).u(\mathbf{r})$ are sound pressure and horizontal particle velocity, $c(\mathbf{r}),\rho(\mathbf{r})$-sound speed and density, ω-angular frequency and $\triangle_{yz},\nabla_{yz}$ are laplacian and gradient in the plane (y,z), are an exact equations for the $P(\mathbf{r})$ and $u(\mathbf{r})$, derived from Euler equations of small motion of compressible fluid. They have first order with respect to x and ready for different marching type algorithms like split-step and Crank-Nicholson, without any approximations of operators, unlike traditional PE scheme. Different examples, illustrating applications of this equations are presented.

1:45

2pUW2. Use of Galerkin's method using variable depth grids in the parabolic equation model. William Sanders (Seafloor Sciences, Naval Research Laboratory, Naval Research Laboratory, Stennis Space Center, MS 39529, wsanders@nrlssc.navy.mil)

Choice of a depth grid affects the accuracy of a parabolic equation (PE) propagation model, as well as the speed of execution. Choice of a fine grid step may result in lower discretization errors, but will lengthen computation

times. Moreover, application of a sampling requirement (e.g. N samples per wavelength) to the water column results in oversampling in the bottom. Since an artificial absorbing later in the bottom is often employed, a large part of the computational domain is oversampled. Fine depth sampling is only needed in the region about the water sediment interface. Hence, Galerkin's method using a variable depth grid is implemented. This is demonstrated to achieve the same error as a uniform grid with fine spacing over the entire depth domain, while taking a fraction of the run time. This is particularly important in models where nested PE models are required, as with noise models (so called N by 2D runs), full 3D, or broadband models. The variable grid Galerkin's method is also used in an elastic PE model, in which sampling requirements for low-shear speed sediments require much finer sampling than that in the water column.

2:00

2pUW3. Applications of a higher order operator splitting scheme on parabolic-equation methods for modeling underwater sound propagation. Ying-Tsong Lin, Timothy F. Duda (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543, ytlin@whoi.edu), Jon M. Collis (Applied Mathematics & Statistics, Colorado School of Mines, Golden, CO), and Arthur E. Newhall (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA)

A higher order operator splitting scheme is presented to increase the accuracy of parabolic-equation (PE) methods. This splitting scheme is essentially applied to the square-root Helmholtz operator resulting from the PE approximation, and it can benefit both the split-step Fourier and split-step Padé methods for modeling sound propagation in three-dimensional environments. Note that the square-root Helmholtz operator in the split-step Fourier PE method is split into a free propagator and a phase anomaly, but in the split-step Padé PE method it is split into two one-dimensional transverse derivatives. Our higher order scheme can handle these two different types of operator splittings, and accuracy improvements come from the fact that our computation includes (to second order) cross terms of the split operators. Numerical underwater acoustic examples are provided to demonstrate the performance of this scheme and comparisons against other schemes. [Work supported by the Office of Naval Research.]

2:15

2pUW4. Propagation modeling of under-ice transmissions using the parabolic equation. Kevin D. Heaney, Richard L. Campbell (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), and Lee F. Frietag (Woods Hole Oceanographic Institution, Woods Hole, MA)

A recent reformulation of the RAM Parabolic Equation model (OASIS' CRAM model) permits efficient Nx2D propagation modelling, including ACOMMS performance estimation. This model is currently set up as a 3-layer propagation model. The 3-layers commonly used are water-column, sediment and basement - where the water-column and sediment have a depth dependent compressional speed, density and attenuation. The basement is an acoustic half-space with a sponge. To model under-ice propagation, the 3-layers can set up as sea-ice, water-column and seafloor. The only loss of generality is the single half-space seafloor, although bottom interaction is not as important a feature in the iso-thermal, upward refracting sound speed profiles in northern latitudes. The ice is modeled as an iso-speed (1700 m/s, rho = .988 kg/m3, alpha= 0.3dB/lambda) fluid on-top of the seawater. The ice-water boundary is input into the code the same way a complex bathymetric profile is used. Although this approach doesn't include shear propagation - an important source of frequency dependent attenuation, it does accurately model the scattering within the water column induced by interactions with the complex under-ice morphology. Comparison of measurements with models will be made for 900 Hz broadband transmissions to ranges of 50 km.

2:30

2pUW5. Seismo-acoustic propagation near thin and low-shear speed ocean bottom sediments. Jon M. Collis (Applied Mathematics and Statistics, Colorado School of Mines, 1500 Illinois Street, Golden, CO 80401, jcollis@mines.edu), Adam M. Metzler (Applied Research Laboratory, University of Texas, Austin, TX), and William L. Siegmann (Mathematical Sciences, Rensselaer Polytechnic Institute, Troy, NY)

Accurate and efficient parabolic equation solutions exist for complex propagation environments with elastic sediments. Certain ocean acoustic environments (such as harbors or estuaries) can feature a seafloor interface consisting of partially consolidated sediments, which can be described as a transitional solid. These complex sediments are generally thin, with low-shear wave speeds, and can cause numerical instabilities to arise in parabolic equation solutions. These instabilities make it difficult to obtain accurate solutions. In the low-shear limit, this problem can be treated as a multiple-scale problem. In this talk, such an ocean environment is modeled as a water layer overlying a thin transitional solid sediment layer over an elastic basement. An elastic parabolic equation approach is developed using asymptotic solutions for the displacements in the transitional solid, which are then incorporated into existing seismo-acoustic parabolic equation solutions by explicitly enforcing fluid-solid and solid-solid interface conditions across the transitional-solid interfaces. Solutions are benchmarked against existing elastic parabolic equation and normal mode solutions for accuracy.

2:45

2pUW6. Three-dimensional numerical modeling of sound propagation and scattering in the deep ocean with elastic (shear) bottoms. Ilya A. Udovydchenkov (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Rd., Woods Hole, MA 02543, ilya@whoi.edu), Ralph A. Stephen (Geology and Geophysics, Woods Hole Oceanographic Institution, Woods Hole, MA), Timothy F. Duda, Ying-Tsong Lin (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA), and Daniel Peter (Department of Geosciences, Princeton University, Princeton, NJ)

A major challenge in bottom-interacting ocean acoustics is to include an accurate description of environmental variability in a computationally feasible model. Wave field predictions are often difficult in environments with strong range dependence, with rapid bathymetric variations, with multiple scattering regions, with interface waves at fluid/solid boundaries, and/or with shear waves in the bottom. In this presentation, we are adapting an existing three-dimensional spectral-finite-element code (SPECFEM3D, distributed and supported by the NSF funded program, Computational Infrastructure for Geodynamics), originally developed for global seismology, to bottom interaction problems in underwater acoustics. Numerical results from SPECFEM3D are compared with the range-dependent acoustic/elastic wave propagation model based on the parabolic equation (PE) method, for a 10 Hz acoustic pulse propagating in the deep ocean. The importance of out-of-plane scattering and bottom shear properties on resulting wave fields are investigated. [Work supported by ONR.]

3:00–3:15 Break

3:15

2pUW7. Effects of seismic source and environment parameters on elastic bottom parabolic equation solutions. Scott D. Frank (Mathematics, Marist College, 3399 North Ave., Marist College Mathematics, Poughkeepsie, NY 12601, scott.frank@marist.edu), Jon M. Collis (Applied Mathematics and Statistics, Colorado School of Mines, Golden, CO), and Robert I. Odom (Applied Physics Laboratory, University of Washington, Seattle, WA)

Recently, two-types of elastic self-starters have been incorporated into parabolic equation solutions for range-dependent elastic bottom underwater acoustic problems. These source fields generate parabolic equation solutions that can be used to study development of oceanic T-phases via the process of downslope conversion. More general range-dependence has also been shown to scatter elastic wave energy into water column acoustic modes which then propagate as T-phases. In certain circumstances, sources in the elastic bottom can also cause interface waves at the ocean bottom that

contribute to the ocean acoustic field. Both types of waves can propagate long distances and could be source mechanisms for unexplained acoustic signals recorded near the sea floor and below the ray-theoretic turning point. Parabolic equation solutions will be used to study effects of parameters such as frequency, source location, and source type on T-phase and interface wave generation and propagation.[Work supported by ONR.]

3:30

2pUW8. Modeling low-frequency seismo-acoustic propagation in the Arctic using a parabolic equation solution. Adam M. Metzler (Environmental Sciences Laboratory, Applied Research Laboratories: University of Texas, 10000 Burnet Rd, Austin, TX 78713, ametzler@arlut.utexas.edu), Jon M. Collis (Applied Mathematics and Statistics, Colorado School of Mines, Golden, CO), and William L. Siegmann (Mathematical Sciences, Rensselaer Polytechnic Institute, Troy, NY)

Propagation in Arctic environments is complicated by three-dimensional variations in the waveguide. The sound speed minimum occurs at or near the ice-covered surface, and the upward refracting profile causes long-range propagation to interact repeatedly with the ice cover. Propagation in the Arctic waveguide needs to include an elastic ice cover of variable thickness which may terminate, overlying an ocean layer and an elastic sediment bottom. Parabolic equation solutions are accurate and efficient for elastic layers, although currently, solutions do not exist for Arctic environments. In this paper, elastic parabolic equation solutions will be obtained for range-dependent Arctic waveguides with elastic ice cover and an elastic bottom. Particular interest will be directed to environments where the ice cover terminates. Results will be benchmarked against normal mode and wave number integration solutions.

3:45

2pUW9. Elastic coupled modes for range dependent propagation. Minkyu Park (Geophysics, Korean Polar Research Institute, Incheon, Republic of Korea), Robert I. Odom (Applied Physics Lab, University of Washington, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, odom@apl.washington.edu), Scott D. Frank (Mathematics, Marist College, Poughkeepsie, NY), and Jon M. Collis (Applied Mathematics and Statistics, Colorado School of Mines, Golden, CO)

The seismo-acoustic field in a range dependent fluid-elastic environment can be computed from elastic coupled modes. Range dependence breaks the strict mathematical orthogonality of the modes causing energy to be exchanged among the elements of the modal spectrum. The original theory is from Maupin (1988). Range-dependent propagation effects are illustrated for a 2-D model including a seamount. The mode coupling is directly proportional to the bathymetric slope of the range dependence. While the range-dependent environmental model is composed of vertical slices, no iteration is required to solve for both the transmitted and the reflected seismo-acoustic field, which is found by solving a matrix Riccati equation for the modal reflection and transmission matrices. A particular feature of the elastic coupled modes is the automatic inclusion of the interface Scholte wave modes, which are absent from any all-fluid environmental propagation model. The excitation of the Scholte modes may be important for observed deep shadow zone acoustic arrivals. The coupled mode results will be compared to the results from an elastic parabolic equation (PE). [Work supported by ONR.]

4:00

2pUW10. Effects of tropical cyclones on underwater sound propagation. Arthur E. Newhall, Ying-Tsong Lin (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA 02543, anewhall@whoi.edu), Sen Jan (Institute of Oceanography, National Taiwan University, Taipei, Taiwan), and James F. Lynch (Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, Woods Hole, MA)

Environmental data collected near the continental shelf and the shelfbreak in the Southern East China Sea around Taiwan are utilized to study sound propagation effects of tropical cyclones (typhoons). These data were from the Conductivity-Temperature-Depth (CTD) profiling casts conducted during the Quantifying, Predicting, Exploiting (QPE) Uncertainty experiments in both 2008 and 2009. These CTD surveys provided observations for the normal physical oceanographic conditions in the area and, most importantly, the extreme conditions induced by a tropical cyclone Typhoon Morakot in the summer of 2009. Strong upwelling currents near the shelfbreak in the experiment area were observed after the typhoon past by, and the water-column stratification was changed. A sound speed variation of 10 m/s over 50 km in distance was measured in the upwelling area, and this can produce significant effects on sound propagation. Numerical models using parabolic-equation and ray tracing methods will be presented to demonstrate the underlying physics of the sound field variability. [Work supported by the Office of Naval Research.]

4:15

2pUW11. The effects of mode structure on coherence in shallow water propagation. Jennifer Wylie, Harry DeFerrari, and Felipe Lourenco (AMP, University of Miami, FL 33432, jennie.wylie@gmail.com)

In an ideal shallow water propagation channel the sound field is accurately described by normal modes and the mode structure is predictable with clean separated modes. However the real ocean environment is rarely ideal, and variations in bottom bathymetry and water column sound speed are usually present. When fluctuations are small and on the order of a fraction of an acoustic wavelength the sound field propagation is deterministic and any mode pattern deviations are small and slow and the phase response is linear. Here the propagation is considered phase coherent and spatial and/or temporal averaging will produce gain. When the fluctuations increase to the order of $1/2$ to 1 acoustic wavelength the mode patterns becomes distorted such that slow linear fluctuations in phase can no longer describe the propagation. Here coherence is reduced and in some cases completely lost. A unifying theory will be presented linking mode pattern deviations and coherence. PE models will be used to predict mode structure and individual mode correlation will be computed with the ideal case. These mode correlations will be used to estimate temporal and spatial coherence and the results will be compared with data from three shallow water experiments.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

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R.J. Peppin, Vice Chair ASC S1
Scantek, Inc., 6430 Dobbin Road, #C, Columbia, MD 21045

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 23 October 2012.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

TUESDAY AFTERNOON, 23 OCTOBER 2012 JAY MCSHAN A, 2:30 P.M. TO 3:45 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C.J. Struck, Chair ASC S3
CJS Labs, 57 States Street, San Francisco CA 94114-1401

G.J. Frye, Vice Chair ASC S3
Frye Electronics, Inc., P.O. Box 23391, Tigard OR 97281

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 23 October 2012.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance and comfort.

TUESDAY AFTERNOON, 23 OCTOBER 2012 JAY MCSHAN A, 4:00 P.M. TO 5:00 P.M.

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D.K. Delaney, Chair ASC S3/SC 1
USA CERL, 2902 Newmark Drive, Champaign, IL 61822

M.C. Hastings, Vice Chair ASC S3/SC 1
Georgia Institute of Technology, G.W. Woodruff School of Mechanical Engineering,
126 Love Building, 771 Ferst Drive, Atlanta, GA 30332 0405

Accredited Standards Committee S3/SC 1 on Animal Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note—those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 23 October 2012.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria or aquariums; or free-ranging wild animals.

TUESDAY EVENING, 23 OCTOBER 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, Wednesday, and Thursday evenings 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 Tuesday are as follows:

Acoustical Oceanography	Mary Lou Williams
Architectural Acoustics	Trianon C/D
Engineering Acoustics	Lester Young A
Musical Acoustics	Andy Kirk
Physical Acoustics	Basie A1
Psychological and Physiological Acoustics	Salon 7 Roosevelt
Structural Acoustics and Vibration	Trianon A/B

Session 3aAA

Architectural Acoustics and Noise: Advancements and Best Practices in Instrumentation for Architectural Acoustics and Noise

Matthew V. Golden, Cochair
Scantek, 6430c Dobbin Rd., Columbia, MD 21045

Eric L. Reuter, Cochair
Reuter Associates LLC, 10 Vaughan Mall, Portsmouth, NH 03801

Chair's Introduction—8:00

Invited Papers

8:05

3aAA1. Special tools and procedures for measuring ground vibration. James E. Phillips (Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608, jphillips@wiai.com)

This talk will present some of the tools and procedures used to analyze ground vibration and the propagation of vibration through soils. These tools and procedures are typically used to evaluate and control groundborne noise and vibration associated ground based transportation such as trains.

8:25

3aAA2. Using real sources to measure the acoustical behavior in rooms. Bruce C. Olson (Ahnert Feistel Media Group, 8717 Humboldt Avenue North, Brooklyn Park, MN 55444, bcolson@afmg.eu)

It is common practice to use dodecahedron loudspeakers and balloons as the source for acoustical measurements. The drawback to this of course, is that it means an interruption to the normal performance in order to collect data in the presence of an audience. New techniques are now available for measurements using real acoustic sources without degrading the performance.

8:45

3aAA3. Advancements in instrumentation for source identification. Matthew V. Golden (Scantek, 6430c Dobbin Rd, Columbia, MD 21045, goldenm@scantekinc.com)

In the beginning, acousticians only had their ears to identify sources. Today we have many more advanced instrumentation, even more than the simple intensity probe that we had just a few decades ago. This paper will review several of these new devices. The first will be an Intensity Tracking System that uses machine vision to produce sound intensity maps. The second is an Acoustic Camera that uses 256 microphones in a two dimension array to create real time videos of the sound field overlaid with standard video. The final instrument will be an environmental noise monitor that uses the arrival times at multiple microphones to detect the location of sources in three dimensional space. Real life measurement results will be shown from each instrument. These new tools give acousticians far more tools than they had just a few years ago.

9:05

3aAA4. Applications of mobile computing devices in acoustics. Benjamin Faber (Faber Acoustical, LLC, 931 Valley View Dr, Santaquin, UT 84655, ben@faberacoustical.com)

In the emerging post-PC era, more and more day-to-day computing tasks will be accomplished with mobile devices, such as the iPhone and iPad. Efforts to bring acoustical measurement and analysis tools to mobile devices have already begun. Mobile devices are not only smaller and lighter even than notebook computers, but they typically employ capacitive touchscreen technology, which enables an unprecedented level of interactivity between user and device. The media-centric nature of the current crop of mobile devices also makes them well-suited for acoustics-related applications. Several examples of hardware and software solutions for acoustical measurements with mobile devices will be presented and discussed.

9:25

3aAA5. Airflow resistance—A comparison between international and american test methods. Marek Kovacik (Scantek Inc., 6430c Dobbin Rd, Columbia, MD 21045, kovacikm@scantekinc.com)

This paper presents the results of a study to determine differences between two test methods that purport to measure airflow resistance. Airflow resistance can be used to determine the acoustical absorption characteristics of materials. In ASTM C522-03(R2009) method, the air is supplied at a steady rate and the pressure difference across the test specimen is measured. In the ISO 9053/EN 29053 B method, a low frequency acoustic wave is produced by a moving piston. The pressure change measured inside the testing apparatus is directly related to airflow resistivity. In a recent round robin, sponsored by ASTM, samples of various densities were tested using the ASTM method. Airflow resistance was also measured with a Norsonic 1517 Airflow Resistance Measurement System following the ISO method. Differences between methods, effects of sample orientation, and other causes of uncertainty will be presented.

9:45

3aAA6. Variations in standing-wave impedance tube design and the effect on the resulting data. Bonnie Schnitta and Greg Enenstein (SoundSense, LLC, 46 Newtown Lane, Suite One, East Hampton, NY 11937, bonnie@soundsense.com)

Standing-wave impedance tubes are a practical and common method for estimating the acoustic absorption characteristics of a material. However, the standing-wave impedance tube test contains inherent variability with its design. Impedance tube standards allow for flexibility in tube material and tube dimensions. These variables in the design criteria of the impedance tube can produce disparities between measured absorption coefficients across different impedance tubes as well as when compared to well-established reverberation room data. Consequently, when designing a tube, in order to obtain accurate absorption values, it becomes necessary to optimize a tube, beyond merely preventing cross modes and allowing for a long enough tube to develop a pressure high and low. This study examines the effects of surface interactions in impedance tubes and how varying tube dimensions, most notably the width, will affect wave development.

WEDNESDAY MORNING, 24 OCTOBER 2012 JULIA LEE A/B, 8:55 A.M. TO 11:15 A.M.

Session 3aAB

Animal Bioacoustics: Vocalization, Hearing, and Response in Non-Human Vertebrates I

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

Chair's Introduction—8:55

Contributed Papers

9:00

3aAB1. Temporal coherence in Budgerigars (*Melopsittacus undulatus*). Erikson G. Neilans and Micheal L. Dent (Psychology, University at Buffalo, SUNY, Buffalo, NY 14260, mdent@buffalo.edu)

Auditory scene analysis has been suggested as a universal process that exists across all animals. Relative to humans, however, little work has been devoted to how animals isolate sound sources to create auditory objects. Frequency separation of sounds is arguably the most common parameter studied in auditory streaming, yet it is not the only factor. Elhilali et al. (2009) found that in humans, synchronous tones are heard as a single auditory stream, even at large frequency separations, compared to asynchronous tones with the same frequency separations, which are perceived as two sounds. These findings demonstrate how both timing and frequency separation of sounds are important for auditory scene analysis. It was unclear how animals, such as budgerigars (Melopsittacus undulatus), would perceive synchronous and asynchronous sounds. Budgerigars were tested on the perception of synchronous, asynchronous, and partially overlapping pure tones and budgerigar contact calls. Budgerigars segregate partially overlapping sounds in a manner predicted by computational models of streaming. However, overlapping budgerigar contact calls are more likely to be segregated than pure tone stimuli with the same temporal overlap. These results emphasize the necessity of using complex communication signals when examining complex sound perception processes such as auditory scene analysis.

9:15

3aAB2. Spatial release from electronic clutter masking in FM bat echolocation. Michaela Warnecke, Mary E. Bates, and James A. Simmons (Neuroscience, Brown University, 185 Meeting St, Providence, RI 02912, michaela_warnecke@brown.edu)

For big brown bats, angular separation of target and clutter echoes causes spatial release from clutter masking. Experiments using echoes that are electronically-generated by loudspeakers show that lowpass-filtering of normally masking echoes also causes clutter masking to disappear. Such lowpass-filtering induces amplitude-latency trading, which retards neural response times from clutter echoes at higher frequencies relative to lower frequencies. Countervailing changes in presentation-times of higher frequencies in electronically-generated clutter echoes restores masking. We present new results showing that moving the clutter-delivering loudspeakers to a different azimuth and elevation causes clutter masking to disappear. But, similar to the earlier experiments, the countervailing changes in presentation of higher frequencies reintroduce masking. In the bat's inferior colliculus, FM sounds that mimic broadcasts and echoes evoke ~1 spike per sound at each neuron's best frequency. However, amplitude-tuning is very broad, so bats work in the latency domain instead, to exploit their high acuity for detecting coherence or non-coherence of echo responses. Overall, the results indicate that big brown bats use neuronal response timing for virtually all auditory computations of echo delay, including those involved in clutter rejection derived from echo spectra. [Work supported by ONR and NSF.]

9:30

3aAB3. Dhole (asiatic wild dog) and tapir vocalizations: Whistling in the jungle. David Browning (Physics Department, URI, 139 Old North Road, Kingston, RI 02881, decibeldb@aol.com) and Peter M. Scheifele (Communication Sciences Dept., University of Cincinnati, Cincinnati, OH)

Few land mammals whistle, and then, as with the marmots, usually just a quick alarm signal to visually alert. Yet in the jungle are found two distinctly different examples, both unique in their animal groups, which employ whistling as a means of communication in an acoustically noisy and visually very limited environment. Dholes, commonly referred to as whistling dogs, belong to a pack which typically breaks into smaller groups to hunt, primarily in the daytime. They whistle to keep in contact while trying to surround game hidden in the bush. In contrast, tapirs are solitary herbivores, grazing primarily at night. Both have found that a frequency sweep whistle is an effective means of communication either in the "screeching noise" daytime environment or the "rasping noise" insect dominated darkness.

9:45

3aAB4. Using bond graphs to model vocal production in túngara frogs.
Nicole M. Kime (Biological Sciences, Edgewood College, 1000 Edgewood College Drive, Madison, WI 53711, nkime@edgewood.edu), Michael J. Ryan (Section of Integrative Biology, University of Texas, Austin, TX), and Preston S. Wilson (Mechanical Engineering, University of Texas, Austin, TX)

Male anurans (frogs and toads) produce species-specific advertisement calls. Conceptual models illustrate how anurans produce calls, but quantitative models of vocal production are rare. Consequently, how frogs produce complex or nonlinear signals, and how differences in morphology result in call differences among species, is in many cases unclear. Bond graphs are representations of dynamic physical systems that allow one to model both the hydraulic elements and mechanical oscillators of air-driven vocal production systems. They can be used to model either single components or interactions among components within vocal systems. This paper uses túngara frogs to show how bond graphs can be used to model vocal production. Túngara frogs produce complex calls that contain a species-specific "whine" and a facultative "chuck". Their calls are frequency- and amplitude-modulated, and exhibit nonlinearities. A bond graph model of the túngara vocal folds produces sustained oscillations at a frequency typical of this frog. The complexity of the túngara frog call is explained by modulating the behavior of integrated components within its vocal production system. The bond graph modeling approach can be used to understand how the biomechanics of vocal production underlies the diversity of animal signals.

10:00–10:30 Break

10:30

3aAB5. Inventory size and complexity in the song of the American Robin.
Kayleigh E. Reyes (Linguistics, UNC Chapel Hill, Chapel Hill, NC 27599, kreyes@live.unc.edu)

This paper discusses American Robin morning song and offers a component system to explain its complex structure and discuss the relationship between inventory size and syllable complexity. Previous research has noted that Robin song is combinatorial, meaning Robins are capable of creating different songs by concatenating elements called syllables. For each robin, all unique syllables produced during a 40 minute recording were grouped into a syllable inventory. Inspired by observations linguists have made about human phoneme inventories, a component system was devised that broke each syllable into its smallest parts. These parts were found by using minima in intensity contours to correspond to natural breaks in the robin's utterances. This component system included all unique syllable parts, 7 total, from 11 syllable inventories with between 8 and 24 different syllables. This system showed certain components occurred more often than others and robins with larger syllable inventories had more rare components on average. It also showed each token of the same syllable type had the same component structure. This component system will also be used to test for patterns in sequencing of components in syllable structure to determine where components can occur and constraints on syllable sequencing in overall song structure.

10:45

3aAB6. Use of social sounds by humpback whales (*Megaptera novaeangliae*) in the Western Antarctic Peninsula feeding grounds.
Michelle Klein (College of the Atlantic, 105 Eden Street, Bar Harbor, ME 04609, mklein@coa.edu) and Douglas Nowacek (Marine Science & Conservation, Duke University Marine Lab, Beaufort, NC)

Humpback whales (*Megaptera novaeangliae*) are renowned for the complex structured songs produced by males. A second, relatively understudied area of humpback acoustic communication concerns un-patterned sounds known as "social sounds," produced by both males and females. This paper explores the use of non-song sounds by humpback whales on the Western Antarctic Peninsula feeding ground. To obtain high quality, close range recordings of humpback whale social sounds, digital acoustic tags (DTAGs) were placed on humpback whales exhibiting foraging behavior during the austral autumn of 2009 and 2010. Overall vocalization rates of two types of social sounds, "wops" and "grunts," showed that there was not a significant diurnal pattern in call production, suggesting that perhaps these calls are not used exclusively for foraging on the feeding ground. These results enhance our understanding of this acoustically advanced species, and will also be useful in conservation and management efforts. The acoustic parameters of the calls that were identified can be used in the development of automatic detection algorithms, and the behavioral contexts during call production can assist in interpretation of passive acoustic monitoring research on humpback whales, and potentially other baleen whale species as well.

11:00

3aAB7. Is the ocean really getting louder?
Michael A. Stocker (Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org) and John T. Reuterdahl (Ocean Conservation Research, Mill Valley, CA)

In 1975 Donald Ross indicated a long term trend of low frequency anthropogenic noise increase of 0.55dB/year between 1958 and 1975. This trend in ocean ambient noise levels due to expansion in global shipping has yielded an increase in the ambient noise floor of the ocean that is anywhere from 6dB to 12dB higher than what it was in 1958 (depending on location). What became known as the "Ross Prediction" did not incorporate other anthropogenic sources of noise such as navigation and communication signals, noise from offshore fossil fuel exploration and extraction, and the noises from other marine industrial enterprises. There is a concern that the increase in ambient noise is masking biologically significant sounds, although the evidence for this is still scarce and somewhat speculative. Meanwhile perhaps 90 percent of the biomass of complex vertebrates has been removed from the ocean since 1850 due to industrialized whaling and fishing operations. This paper examines whether the ocean ambient noise floor may have been significantly higher in 1800 than in the 1958 baseline year of the "Ross Prediction," and speculates that ambient noise levels may be less of a biological aggravator than the particular characteristics of a noise source.

Session 3aAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

Martin Siderius, Chair
ECE Dept., Portland State Univ., Portland, OR 97201

Chair's Introduction—11:00

Invited Paper

11:05

3aAO1. The problem of sound propagation through the fluctuating ocean and the interplay between ocean acoustics and physical oceanography. John A. Colosi (Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Monterey, CA 93943, jacolosi@nps.edu)

While the ocean provides a particularly amenable environment for acoustic remote sensing, navigation, and communication, the confounding effects of fluctuations have bedeviled ocean acousticians for decades. To attack this problem the communities of ocean acoustics and physical oceanography became new bedfellows, with the development of ocean acoustic tomography and path-integral/moment-transport theories driven by internal-wave models being major accomplishments for describing low and high-frequency fluctuations. With the advent of relatively inexpensive oceanographic mooring equipment, experiments in the last decades have deployed nearly as many oceanographic instruments as acoustic instruments, thus leading to fundamental discoveries concerning internal tides, solitons, random internal waves and spicy thermohaline structure. These measurements are in turn providing strong constraints on acoustic fluctuation theories and Monte Carlo simulations used to interpret acoustic observations. This presentation will give a review of what has been learned about ocean sound-speed structure and how this information can be better integrated into acoustic fluctuations calculations. It will be shown that acoustic fluctuation theory has developed to the point in which reasonable inversions for internal-wave parameters are now possible.

WEDNESDAY MORNING, 24 OCTOBER 2012 TRIANON A/B, 7:55 A.M. TO 12:00 NOON

Session 3aBA

Biomedical Acoustics and Signal Processing in Acoustics: Measurement of Material Properties Using Wave Propagation Methods

Matthew W. Urban, Chair
Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, Rochester, MN 55905

Chair's Introduction—7:55

Invited Papers

8:00

3aBA1. Acoustic waves in characterizing biological materials: From molecules to soft tissue and bone. A review. Armen Sarvazyan (Artann Laboratories, 1459 Lower Ferry Rd., Trenton, NJ 08618, armen@artannlabs.com)

Various types of acoustic waves: bulk compressional, shear, surface, and numerous modes of guided waves, are used for characterizing structure and composition of biological media. Bulk waves can be used to assess the composition of biological fluids, such as blood, plasma, milk, urine, stomach juice, and biopolymers in solutions and both bulk and shear waves can be used when characterizing soft tissues. While the assessment of skin is primarily performed using surface waves; various modes of guided waves are used to characterize bone and blood vessels. Historically, the problem of acoustic characterization of tissue properties became a hot topic in the early 1970s in conjunction with the emergence and great success of ultrasonography. The idea of remotely assessing the properties of tissue in a region of interest, as seen on an ultrasonic image, attracted the attention of numerous researchers, but until recently and despite intensive studies and great expectations, not many examples of acoustic characterization of tissue properties made a significant impact. Only in

the last decade, this idea started to bring first fruits and is being implemented in commercial devices which are using shear elasticity modulus of tissue as a characterization parameter.

8:20

3aBA2. Abdominal magnetic resonance elastography. Meng Yin (Radiology, Mayo Clinic, 200 First Street SW, Rochester, MN 55905, yin.meng@mayo.edu)

Many disease processes, both focal and diffuse, cause marked changes in cell and tissue mechanical properties. Magnetic resonance elastography (MRE) is a magnetic resonance imaging-based technique for quantitatively assessing the mechanical properties of tissues based on the propagation of shear waves. Multiple studies have demonstrated MRE can be successfully implemented to assess abdominal organs with many potential applications, from detecting diffuse disease processes to characterizing tumors. The first clinical application of MRE to be well documented is the detection and characterization of hepatic fibrosis, which systematically increases the stiffness of liver tissue. In this diagnostic role, it offers a safer, less expensive, and potentially more accurate alternative to invasive liver biopsy. Emerging results also suggest that measurements of liver and spleen stiffness may provide an indirect way to assess portal hypertension. Preliminary studies have demonstrated that it is possible to use MRE to evaluate the mechanical properties of other abdominal structures, such as the pancreas and kidneys. Steady technical progress in developing practical protocols for applying MRE in the abdomen and the pelvis provides opportunities to explore many other potential applications of this emerging technology.

8:40

3aBA3. A comparison of mechanical wave measurement techniques to quantify soft tissue viscoelasticity up to 8 kHz: A phantom study of shear, Rayleigh and Lamb waves. Temel K. Yasar (Mechanical & Industrial Engineering, University of Illinois at Chicago, Chicago, IL), Thomas J. Royston, and Richard L. Magin (Bioengineering, University of Illinois at Chicago, 851 South Morgan St., Chicago, IL 60607, troyston@uic.edu)

Over the past few decades different techniques based on measurement of mechanical wave motion have been developed for noninvasive quantitative measurement and mapping of soft biological tissue shear viscoelastic properties. In this talk we compare two different measurement approaches, three wave types, and several models for quantifying material viscoelasticity up to 8 kHz for a soft tissue phantom material known as ecoflex. Surface waves and Lamb waves are measured using scanning laser Doppler vibrometry (SLDV). Lamb waves and shear waves are measured using magnetic resonance elastography (MRE). Different linear models of viscoelasticity, including Voigt, Maxwell, more generalized and fractional order types, are optimized and compared based on the different experiments. Challenges and limitations of the different techniques and model types, and their adaptation to more complex biological tissue and anatomical structures are discussed. [Work supported by NIH Grants EB012142 and EB007537.]

9:00

3aBA4. Quantifying viscoelasticity of boundary sensitive tissues using mechanical wave dispersion ultrasound vibrometry. Ivan Nenadic, Matthew Urban, Cristina Pislaru (Mayo Clinic, 200 1st Street SW, Rochester, MN 55905, ivandulan@gmail.com), Miguel Bernal (Institut Langevin, Paris, Paris, France), and James Greenleaf (Mayo Clinic, Rochester, MN)

The cardiovascular diseases such as atherosclerosis, coronary artery disease, hypertension and diastolic dysfunction have been associated with arterial stiffening and decreased ventricular compliance. Noninvasive techniques capable of quantifying elasticity and viscosity of cardiovascular tissues could facilitate early diagnosis and improve treatment. Here, we present a technique that uses focused ultrasound radiation force to excite mechanical waves in the tissue of interest and pulse echo to track the tissue deformation. Analysis of tissue deformation as a function of time using Fourier transforms allows calculation of the phase wave velocity dispersion (change of velocity as a function of frequency) of various modes of deformation. Continuum mechanics equations governing the motion of a viscoelastic plate and a tube are used to model the myocardial wall and arteries, respectively. Dispersion equations are derived for the two geometries and fit to the measured velocity dispersion to estimate tissue elasticity and viscosity. ECG-gated in vivo measurements of porcine myocardial and arterial elasticity and viscosity through a heart cycle are reported. The results show that both elasticity and viscosity increase during systole and decrease during diastole in the myocardium and arteries, consistent with underlying physiology.

9:20

3aBA5. An energy functional approach for inverse characterization of material properties in elastodynamics. Wilkins Aquino (Civil and Environmental Engineeing, Duke University, Hudson Hall, Durham, NC 27708, wa20@duke.edu), Manuel Diaz (Civil and Environmental Engineering, Cornell University, Ithaca, NY), and Matthew Urban (Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, Rochester, MN)

We present an inverse problem methodology based on the Error in Constitutive Equations (ECE) approach for the identification of material properties in the context of frequency-domain elastodynamics. In the ECE approach, we define a cost functional based on an energy norm that connects a set of kinematically admissible displacements and a set of dynamically admissible stresses. The set of kinematically admissible displacements is composed of fields that satisfy essential boundary conditions and possess sufficient regularity (i.e. smoothness). The set of dynamically admissible stresses is composed of fields that satisfy conservation of linear momentum and natural (i.e. traction) boundary conditions. The inverse problem is solved by finding material properties along with admissible displacement and stress fields such that the ECE functional is minimized. Experimental data is introduced in the formulation as a quadratic penalty term added to the ECE functional. The talk will focus on the reconstruction of elastic and viscoelastic properties in heterogeneous materials in the context of frequency-domain dynamics. Our findings indicate that ECE methods provide faster and more accurate results than conventional least-squares minimization. We will show numerical and experimental results that demonstrate the salient features of the method.

9:40

3aBA6. Quantitative ultrasound imaging for assessing and monitoring therapy. Goutam Ghoshal, Jeremy P. Kemmerer, Chandra Karunakaran, and Michale L. Oelze (Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, 405 N. Mathews, Urbana, IL 61801, gghoshal@illinois.edu)

Conventional ultrasound, which is routinely used for diagnostic imaging applications, is mainly qualitative. However, novel quantitative ultrasound (QUS) imaging modes are being adapted to quantify tissue properties for diagnosing disease, classifying tissues and monitoring and assessing therapy. Ultrasound is a propagating wave that interacts with a medium as a function of the spatially-dependent mechanical properties of the medium. By analyzing the backscattered wave, various properties of the propagating media can be quantified. QUS techniques based on parameterizing spectral features and envelope statistics of the backscattered signal were used to monitor and assess therapy from high intensity focused ultrasound (HIFU) treatment. QUS parameters were obtained by fitting theoretical models to backscatter coefficients (BSCs) that are estimated from backscattered radiofrequency signals. Additional parameters were estimated by fitting the homodyned K distribution to the statistics of the envelope of the backscattered signal. These parameters can be related to tissue properties and microstructure, such as the shape, size and organization of microstructure. Experimental results will be presented to demonstrate the applicability of QUS imaging to monitor and assess HIFU treatments on mouse mammary tumors.

10:00–10:15 Break

Contributed Papers

10:15

3aBA7. Focused, radially polarized shear wave beams in tissue-like media. Kyle S. Spratt, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Applied Research Laboratories, University of Texas, Austin, TX 78712, sprattkyle@gmail.com)

In the past decade there has been a surge in the optics literature regarding the unique characteristics of focused, radially-polarized light beams. Of particular interest is the existence of a longitudinal component to the electric field in the focal region of the beam, of comparable amplitude to the radial component and yet with a smaller beamwidth [cf. Q. Zhan, Adv. Opt. Photon. 1, 1-57 (2009)]. In the linear approximation there exists a direct analogy between these light beams and radially-polarized shear wave beams in incompressible elastic media, and hence we may interpret the results found in the optics literature as applying to low-frequency shear waves propagating through tissue-like media. When considering nonlinear effects, however, the fact that the gradient of the field experiences a larger gain due to focusing than the field itself implies that the shear wave case is more susceptible to nonlinear behavior than its optical analog. Second-harmonic generation in the focal region of a focused, radially-polarized shear wave beam in a soft solid is investigated. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and by NIH DK070618.]

10:30

3aBA8. Shear wave generation using hybrid beamforming methods. Alireza Nabavizadeh, James F. Greenleaf, Mostafa Fatemi, and Matthew W. Urban (Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St SW, Rochester, MN 55905, nabavizadehrafsanjani.alireza@mayo.edu)

Elasticity imaging is a medical imaging modality that measures tissue elasticity to aid in diagnosis of certain diseases. Shear wave-based methods have been developed to perform elasticity measurements in soft tissue. Hybrid beamforming applies both the conventional spherical and axicon focusing to produce a beam for generating a shear wave with increased depth-of-field so that measurements can be made with a plane-like shear wave. We present our aperture design and beam optimization performed using Field II simulations. We varied the number of elements devoted to spherical and axicon focusing as well as the opening angle used for axicon focusing. We tested hybrid beamforming in three elastic phantoms and an excised kidney. We evaluated the shear wave speed measurements accuracy in the phantoms as well as the depth-of-field for each hybrid beam. We compared our results with those from using beams generated using spherical and axicon focusing. Our results show that hybrid beamforming is capable of producing a long narrow beam that performs well when among the 128 elements of transducer, 48 elements are allocated to each axicon portion and 32 elements for spherical aperture while the angle of the axicon aperture is set less than 20.

10:45

3aBA9. On the compressional-shear coefficient of nonlinear elasticity in soft isotropic solids. Bojan Guzina, Egor Dontsov (Civil Engineering, University of Minnesota, 500 Pillsbury Drive SE, Minneapolis, MN 55455, guzina@wave.ce.umn.edu), Matthew Urban, Randall Kinnick, and Mostafa Fatemi (Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, Rochester, MN)

Acoustoelasticity is a technique that allows quantification of the elastic nonlinearity coefficients (the so-called third-order moduli) of a material by measuring the variation of the sound speed in different directions (and/or polarizations) versus the applied uniaxial static stress. When dealing with nominally isotropic solids, the variation of the shear wave speed in two orthogonal directions with the applied stress permits the computation of two out of three independent nonlinearity coefficients. To generate the shear wave in an experimental setting, the acoustic radiation force pulse was applied to uniaxially deformed phantoms with strains of up to 30%. The results demonstrate that the compressional-shear coefficient C, which governs the variation of the (linear) shear modulus with hydrostatic pressure was found to vary substantially from one phantom to another, with the highest value observed in an agarose-based phantom. The importance of this nonlinearity parameter resides in the fact that the magnitude of the acoustic radiation force (ARF) in soft solids is proportional to C-1. For consistency, the values of C obtained from the acoustoelasticity experiment are compared to those deduced from the shear wave amplitude via its relationship to the magnitude of the ARF used to generate the shear wave.

11:00

3aBA10. Use of the radon transform for estimation of shear wave speed. Matthew W. Urban and James F. Greenleaf (Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First Street SW, Rochester, MN 55905, urban.matthew@mayo.edu)

Many methods in the field of elasticity imaging use shear waves to investigate the material properties of various soft tissues. The accurate and robust measurement of the shear wave speed is necessary for reliable clinical measurements. We propose using the Radon transformation on the spatiotemporal shear wave data to estimate the shear wave speed. A similar algorithm called the Radon sum transformation was proposed by Rouze, *et al* (IEEE Trans. Ultrasonics Ferroelectr. Freq. Control. 2010. pp. 2662-70), but this algorithm requires different input parameters that can affect the results. The use of the full Radon transform allows for the differentiation of waves that are traveling in different directions, thus this method can be used as a directional filter. We will also demonstrate the connection between estimating shear wave speeds in the Radon transform domain and estimating shear wave speeds from Fourier transform k-space. Results from shear wave measurements using shear waves induced by ultrasound radiation force will be shown. We will examine the accuracy of measurements made in calibrated elasticity phantoms, and show examples of shear wave speed

estimation in arteries and kidneys. [This research was supported in part by NIH grants EB002640 and DK082408.]

11:15

3aBA11. Radiation-force-based estimation of acoustic attenuation using harmonic motion imaging. Jiangang Chen, Gary Y. Hou, Fabrice Marquet, and Elisa Konofagou (Department of Biomedical Engineering, Columbia University, Columbia University Medical Campus, 622 W 168th, New York City, NY 10032, ek2191@columbia.edu)

Tissue characterization such as attenuation estimation remains challenging but important. Attenuation represents the energy loss during wave propagation through biological tissues, thus affects both therapeutic and diagnostic ultrasound applications. In this study, a novel attenuation estimation approach was developed using radiation-force-based method of Harmonic Motion Imaging (HMI). The HMI set-up comprised of a forcing transducer (fcenter = 4.7 MHz, AM frequency = 25 Hz) in conjunction with a confocal pulse-echo transducer (fcenter = 7.5 MHz), with the former inducing tissue displacement and the latter simultaneously acquiring RF signals. Tissue displacements were estimated from the RF signals using a 1-D cross-correlation method (window size: 1 mm; overlap: 90%). 2D displacement images were obtained through raster-scan (10×10 mm^2). A linear regression model was applied to the displacements at different depths for calculating attenuation. Gel phantoms with known attenuation (n = 5) (CIRS Inc.) and in vitro canine livers (n = 3) were tested. Results demonstrated that attenuations obtained from the phantoms showed good correlation ($R^2 \approx 99\%$) with the independently obtained values (0.28–1.45 dB/cm/MHz) (13 $\pm$ 8% underestimated), while those from the canine liver were 0.32 $\pm$ 0.03 dB/cm/MHz, within the normal range reported in the literature (0.28–1.01 dB/cm/MHz) (Duck, Academic Press 1990). Future studies will entail attenuation measurements in pathological tissues and HIFU monitoring.

11:30

3aBA12. Elastography and tactile imaging for mechanical characterization of superficial muscles. Diego Turo (Department of Bioengineering, George Mason University, Fairfax, VA), Paul Otto (Dept. of Electrical and Computer Eng., George Mason Univ., Fairfax, VA), Vladimir Egorov, Armen Sarvazyan (Artann Laboratories, Trenton, NJ), Lynn H. Gerber (Department of Rehabilitation Science, George Mason University, Fairfax, VA), and Siddhartha Sikdar (Dept. of Electrica and Computer Eng. and Bioengineering, George Mason Univ., 4400 University Drive, MS 1G5, Fairfax, VA 22030, ssikdar@gmu.edu)

Quantification of the mechanical properties of muscle is of significant clinical interest. Local changes in the mechanical properties of muscle are often associated with clinical symptoms. In particular, myofascial trigger points (MTrPs) are a very common, yet poorly understood and overlooked, cause of nonarticular musculoskeletal pain. MTrPs are localized, stiff, hyperirritable tender nodules, palpated in taut bands of skeletal muscle. Objective validated measures of the mechanical properties of MTrPs could potentially be a clinical outcome measure. We are investigating ultrasound shear wave elastography and tactile imaging as complementary objective methods to assess the mechanical properties of MTrPs. In an ongoing clinical study, we recruited 50 subjects (27 healthy controls and 23 with symptomatic chronic neck pain and active MTrPs). The upper trapezius muscles in these subjects were imaged using shear wave elastography using an external vibration source with varying frequency in the range [50–200] Hz to measure shear wave speed and dispersion in tissue, and tactile imaging using an array of pressure sensors allowing 3D reconstruction of mechanical structure of tissue. Preliminary analysis demonstrates that symptomatic muscle tissue in subjects with neck pain is mechanically more heterogeneous and stiffer compared to normal muscle in control subjects ($p < 0.05$).

11:45

3aBA13. Rayleigh wave propagation method for the characterization of viscoelastic properties of biomaterials. Siavash Kazemirad and Luc Mongeau (Mechanical Engineering Department, McGill University, 817 Sherbrooke Street West, Montreal, QC H3A 0C3, Canada, siavash.kazemirad@mail.mcgill.ca)

The frequency-dependent viscoelastic properties of injectable biomaterials used for vocal fold augmentation and repair must be characterized to ensure the integrity with the vibrating tissue throughout the frequency range of vocalization. Experimental methods for quantifying the frequency-dependent viscoelastic properties of biomaterials over a broad frequency range (i.e., up to 4 kHz) using Rayleigh wave propagations were investigated. Appropriate models for Rayleigh wave propagations in single and layered media were developed. Different silicone rubber samples were made and tested to evaluate the proposed methods. Rayleigh waves at different frequencies were launched on the surface of different samples; i.e., single layer samples and samples composed of a substrate with known material properties coated with a thin layer of the soft material that is to be characterized. The input vibrations of the actuator and the motion of the sample surface were measured using an accelerometer and a laser Doppler vibrometer, respectively. A transfer function method was used to obtain the complex Rayleigh wavenumbers. Finally, the complex shear and elastic moduli and the loss factor of samples were calculated through the proper modelling using the measured wavenumbers. The results were compared and shown to be in good agreement with those obtained from other measurement methods.

Session 3aEA

Engineering Acoustics and ASA Committee on Standards: Sound Intensity Measurements

Allan J. Zuckerwar, Cochair
Analytical Services & Materials, 1052 Research Dr., Hampton, VA 23666-1340

Robert J. Hickling, Cochair
Sonometrics Inc., 8306 Huntington Rd., Huntington Woods, MI 48070-1643

Invited Papers

8:00

3aEA1. Sound-power flow. Robert J. Hickling (Sonometrics Inc., 8306 Huntington Road, Huntington Woods, MI 48070-1643, hickling.robert@gmail.com)

To understand sound-intensity measurement, it should be realized that sound intensity is sound-power flow per unit area in the direction of sound propagation. Sound intensity can be measured in fluids such as air or water. Generally measurements cannot be made inside a solid. However there may be ways of measuring sound intensity in a solid, if the solid is transparent. Also sound intensity can be computed in a solid and then measured in a surrounding fluid to collate with the calculated results. In general the measurement of sound-intensity is relatively new. It can be measured either as a vector in three or two dimensions or as a single vector component. These features are discussed, together with some examples of sound-intensity measurement. An important use of a single component of sound intensity is the measurement of the sound power of a noise source. Another use is locating the primary sources of noise in operating machinery. The full sound-intensity vector can determine the direction of a noise source. Two vectors can determine the location of the source in space.

8:25

3aEA2. Complex technique of sound intensity measurements and properties of the basic sound fields. Jiri Tichy (Acoustics, Penn State Univ., State College, PA 16804, tichy@engr.psu.edu) and Gary W. Elko (President, mhacoustics LLC, Summit, NJ)

An overview of sensors and signal processing for the measurement of sound intensity and other energy related quantities such as reactive intensity, potential and kinetic energy is summarized. Many examples of energy propagation, vortices formation, radiationn from sources, sound interference and other acoustic phenomena are presented and analyzed.

8:50

3aEA3. Sound intensity measurements in vehicle interiors. Svend Gade (Brüel & Kjær University, Skodsborgvej 307, Nærum DK-2850, Denmark, sgade@bksv.com), Jørgen Hald, and Jakob Mørkholt (Brüel & Kjær University, Brüel & Kjær Sound & Vibration Measurement, Nærum, Sjælland, Denmark)

In some cases it is important to be able to measure not only the total sound intensity on a panel surface in a vehicle cabin, but also the components of that intensity due to sound radiation and due to absorption from the incident field. For example, these intensity components may be needed for calibration of energy flow models of the cabin noise. A robust method based on surface absorption coefficient measurement is presented in this paper.

9:15

3aEA4. Wide band pressure and velocity (p-v) tympanometry with calibrated sound intensity micro-probes. Domenico Stanzial and Giorgio Sacchi (Research Section of Ferrara, CNR - Institute of Acoustics and Sensors “Corbino”, v. Saragat, 1, Ferrara, Ferrara 44122, Italy, domenico.stanzial@cnr.it)

Wide band p-v tympanometry can be defined as the measurement of the acoustic immittance of the ear, possibly in normal air pressure condition of the ear canal, and in the full audio frequency range. The most important innovation pioneered by the p-v tympanometry regards the introduction of a different principle of measurement based on the direct acquisition of, both, pressure and velocity (p-v) signals at the ear canal entrance. The measurement can be done by using a pre-calibrated couple of dedicated micro-sensors: an ordinary microphone and an acoustic velocimetric micro-device. This invited speech will report about wide band measurements of ear immittance functions carried out by means of a modified tympanometric probe hosting a pre-calibrated sound intensity micro-probe, and their comparison, with data obtained by standard 226 Hz tympanometry.

9:40

3aEA5. A comprehensive examination of the acoustic vector fields scattered by cylindrical bodies. Robert J. Barton, Geoffrey R. Moss (Naval Undersea Warfare Center Division Newport, 1176 Howell St, Newport, RI 02841, robert.barton@navy.mil), and Kevin B. Smith (Naval Postgraduate School, Monterey, CA)

In this study, the properties of the scattered acoustic vector fields generated by infinite-length and finite-length rigid and elastic cylinders are investigated. Analytical solutions are derived from general acoustic pressure scattering models, and analyzed for wave numbers in the resonance region. The separable active and reactive components of the acoustic intensity are used to investigate the structural features of the scattered field components. Numerical results are presented for the near field and transition regions. A finite element model is developed for both rigid and elastic cylindrical bodies. The finite cylinder model and analysis is then extended to include interactions with an elastic half space. The vector properties of the time-independent complex intensity components and their relations to field energy density quantities are summarized.

10:05–10:20 Break

10:20

3aEA6. Investigating measurement of acoustic intensity for rocket sound field characterization. Kent L. Gee (Dept. of Physics and Astronomy, Brigham Young University, N243 ESC, Provo, UT 84602, kentgee@byu.edu), Jonathan D. Blotter (Dept. of Mechanical Engineering, Brigham Young University, Provo, UT), Scott D. Sommerfeldt, Derek C. Thomas (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT), Kenneth S. Bostwick (Dept. of Mechanical Engineering, Brigham Young University, Provo, UT), and Benjamin Y. Christensen (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT)

An effort to characterize the aeroacoustic source regions and noise environment around launch vehicles has resulted in study of the hardware and processing methods used to calculate acoustic intensity. Because of the extremely harsh measurement environment and other source region characteristics, these investigations have included selection, calibration, and arrangement of microphones and examination of the required pressure and particle velocity estimates. The results of analytical, laboratory, and field experiments are described as a summary of lessons learned during the on-going effort.

10:45

3aEA7. A comparison of directional robustness for endfire versus Blumlein microphone arrays used in hearing aids. Thomas Burns (Starkey Hearing Technologies, 6600 Washington Ave S, Eden Prairie, MN 55416, tom_burns@starkey.com)

An endfire microphone array uses two omnidirectional microphones in a delay-and-sum configuration. A Blumlein array mixes one omnidirectional and one (bi)directional mic. Each can be engineered to provide any 1st order directional pattern. The three critical factors for providing good directionality include the relative sensitivity and phase between the microphones in addition to the placement of the hearing instrument on the user's head. In this context, a directional system is robust if its factors can operate over a wide range of levels without degrading the directional performance. In this study, each array was engineered to have the same aperture spacing and tuned to the same freefield polar pattern; this tuning provided the nominal operating levels. Both arrays were placed in-situ on a measurement manikin and 614 impulse responses were acquired in ten degree resolution on all four microphones for different in-situ positions. The data for each array were combined as described above, and the aforementioned factors were varied around the nominal range of levels in a simple central composite design of experiments. The results of the in-situ directional response show improved robustness for a Blumlein mix.

11:10

3aEA8. Doubly steered array of modal transducers. John L. Butler, Alexander L. Butler, and Michael J. Ciufo (Image Acoustics, Inc, 97 Elm Street, Cohasset, MA 02025, jbutler@imageacoustics.com)

Line or planar arrays steered to end fire generally require quarter wavelength spaced transducer elements with 90 degree sequential phase shifting to attain a unidirectional beam in an end fire direction. Half wave length element separation with 180 degree sequential phase shifts yield end fire but in both directions and at a reduced level. Part of this lower level bipolar directionality is due to the fixed broadside directionality of the transducer elements. However, use of reduced-size, leveraged-circular, acoustical-modal transducer elements, which allow incremental steering, provide a means for substantial end fire steering without the need for quarter wavelength spaced elements. In this case the elements, as well as the array, can be steered in the same direction attaining full-strength, unidirectional end fire steering at half wavelength array spacing. We present the physics of the leveraged-circular, acoustic-modal, transducers and their operation in the monopole, dipole and quadrupole modes along with their implementation and application in doubly steered arrays.

11:35

3aEA9. Underwater vector intensity measurements in the ocean and laboratory. David R. Dall'Osto and Peter H. Dahl (Mechancial Engineering and the Applied Physics Laboratory, University of Washington, Seattle, WA 98103, dallosto@u.washington.edu)

Underwater measurements of the acoustic intensity vector field can be provided by either spatially separated hydrophones or by a sensor measuring a property of particle motion, such as particle acceleration. These measurements are used to formulate the vector intensity as the product of pressure and particle velocity. The magnitude of the vector intensity is not necessarily equal to the plane-wave intensity (the mean square pressure divided by the density and sound-speed of the medium) which is often used to define pressure measurements in terms of intensity. In regions of strong destructive interference, the magnitude of the vector intensity may be greater than the plane-wave intensity. Measurements of an impulsive source on a vertical line array of pressure sensors spanning a shallow sea (60 m) off the coast of South Korea are presented to demonstrate properties of the complex intensity vector field in an ocean waveguide. Here, the vertical complex intensity is formulated by finite-difference methods. These vertical intensity observations in the ocean waveguide have implications on properties of the complete vector field. A laboratory experiment using a tri-axial particle acceleration sensor is presented to provide a connection between measurement of elliptical particle motion and complex intensity.

Session 3aEDa

Education in Acoustics: Hands-On Acoustic Demonstrations for Middle-School Students

Andrew Morrison, Chair
Joliet Junior College, Natural Science Dept., Joliet, IL 60431

Approximately 20 acoustics demonstrations will be set up for local students to interact with at the meeting. These students will be assisted by Acoustical Society of America (ASA) Education in Acoustics members and the Student Council members. Conference participants are encouraged to attend this session to help guide student exploration in acoustics phenomena.

WEDNESDAY MORNING, 24 OCTOBER 2012 BASIE FOYER, 10:00 A.M. TO 12:00 NOON

Session 3aEDb

Education in Acoustics: Undergraduate Research Exposition Poster Session

Mardi C. Hastings, Cochair
George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405

Preston S. Wilson, Cochair
Applied Research Lab., Univ. of Texas at Austin, Austin, TX 78712-0292

Contributed Papers

All posters will be on display and all authors will be at their posters from 10.00 a.m. to 12.00 noon.

3aEDb1. Effect of boundary diffusers in a reverberation chamber: A preliminary investigation. Jacob R. Adelgren, David T. Bradley (Physics + Astronomy, Vassar College, 124 Raymond Avenue, #745, Poughkeepsie, NY 12604, dabradley@vassar.edu), Markus Mueller-Trapet, and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Aachen, Nordrhein-Westfalen, Germany)

In this project, the sound field behavior in a 1:5 scale reverberation chamber has been measured and analyzed. Both hanging diffusers and boundary diffusers have been applied in an effort to increase the chamber's sound field diffusivity, which has been characterized based on the guidelines set forth in several American and international standards, including ASTM C423, ASTM E90, and ISO 354. Objective data from measured impulse responses for several configurations of the diffusers will be presented. These data will be compared to those from the empty chamber and to the criteria from the standards. The relative effectiveness of hanging diffusers vs. boundary diffusers will be discussed.

3aEDb2. Source tracking and scatter localization in a reverberant environment. Laura M. Williamson, Justin D. Risetter, Michael A. Pierfelice, and David R. Dowling (Dept. of Mech. Engineering, University of Michigan, Ann Arbor, MI 48109, drd@umich.edu)

Matched field processing (MFP) has been shown to be effective for remote sound source localization when the receiving array clearly records direct-path sound from a stationary source. Unfortunately, in imperfectly characterized confined environments, source motion, echoes, and reverberation commonly degrade localization performance. This poster presentation describes three acoustic technology development efforts focused on using matched-field processing, with and without the first reflections, (i) to track a moving source, (ii) to improve localization results by adjusting the receiving array geometry, and (iii) to determine the conditions under which a discrete scatterer may be localized. Experiments were conducted in a 1.0-meter-deep and 1.07-meter-diameter cylindrical water tank using a single sound projector, a receiving array of 16 hydrophones, and a linear actuator capable of moving the source at a speed of 0.5 m/s. Measured localization performance is reported for impulsive (100 micro-second) and longer duration signals having center frequencies from 30 kHz to near 100 kHz. As expected, source and scatterer localization accuracy is found to be limited by reverberation. The eventual application of this research is localizing sub-visual cavitation bubbles and other hydroacoustic sound sources in hydrodynamic test facilities. [Work supported by NAVSEA through the Naval Engineering Education Center.]

3aEDb3. Doppler measurement of the motion of a physical pendulum. Jean Paul Ngabonziza and Carl Frederickson (Physics and Astronomy, University of Central Arkansas, Conway, AR 72035, carlf@uca.edu)

The Doppler shift of a reflected acoustic signal has been used to characterize the motion of a physical pendulum. The pendulum is covered with a rough surface to provide specular reflection at any angle. Comparison between theoretical and measured spectrograms will be presented. The measurement dependence on the frequency of the source signal will be explored. Source frequencies will be in the audible range. The system is being evaluated for use with a double physical pendulum modeling the motion of a human leg.

3aEDb4. Auditory change detection with common and uncommon sounds. Amanda L. Heberle and Caroline M. DeLong (Psychology, Rochester Inst. of Tech., 6750 Lakeside Rd., Ontario, NY 14519, amanda.heberle@gmail.com)

Change deafness, the inability to detect a change in an auditory scene, is similar to change blindness, the inability to detect a change in a visual scene. In this experiment, participants were asked to detect changes in auditory scenes (one, three, or five sounds). The sounds were either common sounds (e.g. alarm clock) or uncommon sounds (e.g. science fiction laser). Only one sound was modified in pitch or loudness for half of the trials. Participants were not always able to detect changes in a sound sequence (M = 67.1%) even though they could almost perfectly discriminate between the ten sounds (M = 99.2%). Participants performed best with a scene size of one (M = 82.6%) and worse with a scene size of five (M = 63.8%). Participants performed significantly better with common sounds (M = 74.7%) vs. uncommon sounds (M = 69.1%). Participants were significantly better at detecting a pitch change (M = 80.8%) than a loudness change (M = 53.5%). These results are consistent with the idea of change deafness. We remember the gist of an auditory scene but we don't detect changes in every sound.

3aEDb5. Techniques for measuring ultrasonic tissue properties of cells. Aislinn R. Daniels, Aditya D. Mahajan, Yim J. Rodriguez, and Maria-Teresa Herd (Physics, Earlham College, 801 National Road West, Richmond, IN 47374, herdma@earlham.edu)

Measuring ultrasonic characteristics of cells outside the cellular matrix is of interest. Analyzing these properties at the cellular level may identify qualities specific to a cell type, possibly leading towards cell identification and disease diagnosis without invasive techniques such as biopsy. The purpose of this research is to develop a reliable method of examining cellular characteristics using quantitative ultrasound. Measurements were made using single element transducers at frequencies of 5-50 MHz in a controlled water-tank environment. Speed of sound and attenuation were measured using through transmissions with unfocused transducers, and backscatter was measured using pulse/echo transmissions with focused transducers. To test our experimental techniques we measured high-frequency properties of a tissue mimicking phantom and compared the results to the current standards. As our experiment required testing at a smaller scale than previous tests of these methods, we also created small holding tubes with smaller phantoms of the same material to compare the larger sample measurements. These miniature phantoms show a remarkable consistency in the data obtained when compared to a large phantom, which verifies the applicability of the methods on a small scale.

3aEDb6. Ultrasound characterization of Chinese hamster ovary cells. Aditya D. Mahajan, Aislinn R. Daniels, Yim J. Rodriguez, and Maria-Teresa Herd (Physics, Earlham College, 801 National Road West, Richmond, IN 47374, herdma@earlham.edu)

Ultrasonic characteristics of various tissues are currently not known in enough detail to be used reliably for tissue identification or diagnosis. Analysis at a cellular level as opposed to a tissue level can examine these qualities specific to a cell type. The purpose of this research is to find the ultrasonic tissue characterization of Chinese hamster ovary (CHO) cells to develop a general test for modeling cells. To analyze the characteristics, CHO cells are cultured and prepared into a pellet-sized sample, which are then scanned with single element transducers at high frequencies (5–50 MHz). The speed of sound and attenuation of the pellets are measured using through transmissions with unfocused transducers, and backscatter coefficients are measured using pulse/echo transmissions with focused transducers. This study may establish a possible model and experimental method, in addition to providing a control for the characteristics of other cell types, specifically comparing normal and cancerous cells.

3aEDb7. Quantitative ultrasound characterization and comparison of prostate cancer cells and normal prostate cells. Yim J. Rodriguez, Aislinn R. Daniels, Aditya D. Mahajan, and Maria-Teresa Herd (Physics, Earlham College, 801 National Road West, Richmond, IN 47374, herdma@earlham.edu)

Ultrasound plays an important role in helping diagnose prostate cancer as part of Ultrasound Guided Biopsies; however by better characterizing normal and cancerous prostate cells - and not the actual tumor- this study enhances ultrasound as a first-hand diagnostic tool. Using quantitative ultrasound, normal and cancerous prostate cells were analyzed and compared. Experiments to determine tissue characteristics were performed using single element transducers ranging from 5-50 MHz. Measurements of speed of sound, attenuation, and backscatter coefficients were made. The current results present a valuable insight on the differences between benign and malignant formations by analyzing them at the cellular level. Analysis of cellular behavior at smaller scales provides significant information for better understanding the properties of tumors at a larger scale. These findings contribute to enhance tissue characterization. Moreover, the results obtained present relevant data regarding speed of sound, attenuation, and backscatter coefficients useful for comparative studies and further analysis.

3aEDb8. Designing and building transducers for use in a molecular acoustics experiment. Ashley J. Hicks and William V. Slaton (Physics & Astronomy Department at the University of Central Arkansas, 201 Donaghey Ave, Conway, AR 72035, a.jean.hicks@gmail.com)

This work describes the design, construction, and testing of two capacitance transducers for use in a larger project investigating the molecular absorption of sound in certain gases. The transducers are based on designs presented in the literature, modified to work optimally in our system which consists of 4-inch diameter steel pipe. The experiments will be conducted at atmospheric pressure, eliminating design constraints involved when using high pressure gas. However, work done by Bass & Shields shows that to work in these experiments at atmospheric pressure, transducers must have a frequency range of 1 kHz - 100 kHz. [J. Acoust. Soc. Am. Vol 62, p. 346-353, 1977] The basic concept of our transducer depends upon creating a parallel plate capacitor from metal that is flexible enough to move when a sound wave hits it. Our design utilizes 0.051 mm thickness alumninized Mylar film tensioned with a brass retaining ring over a brass backing plate with both secured to a Delrin plastic base for its electrically insulating properties. We will report on the transducer's frequency response characteristics and initial testing in a send/receive configuration in a sound absorption experiment.

3aEDb9. The role of vowel inherent spectral change in the intelligibility of English vowels spoken by English-, Chinese-, and Korean-native speakers. Stephanie Tchen, Su-Hyun Jin, and Chang Liu (Communication Sciences and Disorders, University of Texas, 1 University Station, Austin, TX 78712, shjin@mail.utexas.edu)

This study is designed to investigate the relationship between Vowel Inherent Spectral Change (VISC) and vowel intelligibility of native and non-native speakers of English. VISC refers to the relatively slow varying changes in formant frequencies associated with each vowel category (Neary and Assman, 1986). Such spectral change has been known to be a major factor in the perception of both the phonetic and phonemic English vowels. In the previous research projects conducted in our lab, we recorded 12 English vowels in /hVd/ format spoken by English native (EN), Chinese native (CN) and Korean native (KN) speakers and examined vowel intelligibility. Overall, vowel intelligibility was significantly higher for native talkers than for non-native talkers. Within non-native speaker groups, CN had slightly higher intelligibility than KN speakers. In this study, we are going to analyze an acoustic feature of the vowels, VISC, spoken by these native and non-native speakers. It is hypothesized that the acoustic differences of vowels spoken by different groups of speakers can account for the variances in vowel intelligibility.

3aEDb10. Floor vibration response to Irish dancing. Valerie K. Smith and Murray S. Korman (Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

Using Irish step dance impulses of actual techniques, one could use various vibration sensors (B&K microphone and I/O SM11 geophone) to perform a time frequency analysis of the transient response of a supported portable wooden dance floor resulting from forced transient vibration. The steps included (1) a "tap" (the wooden tap on the toe of the shoe hitting the floor), (2) a "stamp" (a combination of the wooden toe and plastic heel hitting the floor simultaneously) and (3) a "shuffle" (a brushing of the wooden tap on the toe once forwards and once backwards against the dance floor). Experiments were performed using laminated veneer lumber (plywood)

supported by four small rubber mounts near the edges. Floors were (a) 1 m square (d = 3/4 inch thick), (b) 0.5 m square (d = 1 inch), (c) 1m by 0.5m (d = 1 inch) and (d) 0.5 m diam (d = 1 inch). FFT analysis of a transient is compared with the geophone/microphone frequency response (same location) using a swept sine loudspeaker excitation. In (b) the lowest frequencies were 110 and 470 Hz for a "tap" at the center. Performance is enhanced. Green's function analysis is presented. [Ji, and Ellis, The Structural Engineer 3rd ser. 72 (1994), p 37- 44.]

3aEDb11. Nonlinear scattering of crossed ultrasonic beams in a constricted flow for the detection of a simulated deep vein thrombosis: Part II. Markus S. Rebersak and Murray S. Korman (Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

Experiments show that the turbulence generated by a thrombosis (ping pong ball) lodged into a polyethylene cylindrical tube (leg vein) can be detected by the nonlinear scattering at the combination (3.8 MHz sum frequency) from two mutually perpendicular primary wave components (f1=1.8 MHz and f2 = 2.0 MHz). In study (1) the nonlinear scattering at the sum frequency is measured vs. angle from turbulence generated by a submerged water jet. In (2) the thrombosis model vein is submerged in the turbulent flow field, while in (3) the vein remains in place but the thrombosis is removed. Nonlinear scattering at the combination frequency in (1) shows significant Doppler shift and frequency broadening vs. angle. In (2) nonlinear scattering exists but is diminished in amplitude, Doppler shift and spectral broadening, as was expected. In case (3), there is virtually no scattering at the sum frequency, since the vein mitigates the turbulent flow. Results are presented at fine angular increments, and improved alignments to measure mean Doppler shift, standard deviation, skewness and kurtosis vs. scattering angle and thus characterize certain aspects of the turbulence behind the clot. Results extend the original work of Sean M. Mock [J. Acoust. Soc. Am., 129, 2410 (2011)].

3aEDb12. Electro-dynamic soil-plate-oscillator transducer for monitoring the buried vibration response to airborne sound excitation. Amie E. Nardini and Murray S. Korman (Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

A plastic drum-like anti-personal mine simulant (2 inch diam, by 1 inch tall, by 1/4 inch thick aluminum tube, capped by a 1/4 Al bottom circular plate and an elastic acrylic 1/16 inch thick circular top plate) was constructed. It was then modified to generate an electrical response to top-plate vibration. The mine simulant was converted to a transducer by fastened a sewing machine size bobbin wound with fine enamel copper wire onto the inside face on the bottom plate of the simulant. Next, a rare earth magnet was fastened to the inside surface of the elastic top plate (using a tacky wax). Leads were connected using insulated feed through connectors to complete the transducer design. In testing out the electro-dynamic transducer landmine simulant, results showed that the design was adequate to detect the vibration resonant response using a 2 inch burial depth in a concrete soil box filled with dry masonry sand. Two 12 inch diameter subwoofers were located a meter above the soil box and radiated sound levels on the order of 80-90 dB re 20 micro Pa, were measured near the soil surface. The swept sinusoidal transducer/microphone response exhibited a fundamental resonance near 190 Hz.

3aEDb13. Monitoring the pressure impact of a runner's footstep on the inner sole of the shoe. Jacqueline A. Blackburn and Murray S. Korman (Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

The research goals are to learn about the biomechanics of human footsteps and apply the knowledge to the understanding of the subsequent wave energy that is transferred into the ground or track surface. Phase (1) research is focused on learning about the time development of the footstep pressure that a runner's foot is transferring to the innersole of an athletic shoe, along with accelerometer information. A Tekscan FlexiForce Model A401 force sensor (25.4 mm sensing diameter and 0.208 mm thick) coated with a polyester substrate was chosen due to its versatile force range if biased and placed in an operational amplifier feedback loop. The response time is $<$5 microseconds. Two force sensors will be placed on the upper side of a removable innersole to measure force near the ball and heel of the foot. Phase (2) is to use a Digi Intl XBee 802.15.4 Development Kit to communicate (using a wireless link) the transducer voltage responses. The transmitting module is strapped to the runner's ankle. A receiving module receives data streams and connects to a USB computer interface. Phase (3) converts the data streams to measurements of pressure vs. time. Preliminary experiments and a data analysis will be presented.

3aEDb14. Investigation of helmet-to-helmet collisions in football: Experiments using a mechanical lumped element coupled harmonic oscillator model structure for the skull, fluid, and brain mass. Duncan M. Miller and Murray S. Korman (Physics Department, U.S. Naval Academy, 572 C Holloway Road, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

The study of extreme helmet to helmet football collisions may lead to future precautions to prevent serious head injuries. First, (a) oscillations of the helmet alone (much like a tuning fork) are investigated. Then (b) a thin walled 1/16 inch thick 2 inch tall by 6 inch diameter polycarbonate hoop is used to model the skull (in the n = 2 mode). Next, (c) the hoop is filled with a light weight polyurethane foam to model fluid in the structure. Then (d) a solid brass cylindrical 1 Kg weight is inserted in a carved out slot in the foam. The hoop-foam-brass weight structure is studied in transient vibration. Finally (e) the "skull", "fluid", "brain mass" structure is set in the helmet and cushioned with extraneous foam. A second identical helmet on a pendulum is released with some angular momentum and collides with the helmet (fitted with the structure (e)) that is suspended vertically by its own pendulum cord - initially at rest. In laboratory collision studies three single axis accelerometers are placed on (1) the helmet at rest, (2) the hoop and (3) the end of the cylindrical mass, in an effort to rudimentary model the vibration of the model brain mass.

3aEDb15. Acoustic properties of flight approved materials. Justin Mann, Matthew Sisson, and William Slaton (Physics and Astronomy, University of Central Arkansas, 201 Donaghey Ave, Conway, AR 72035, wvslaton@uca.edu)

The purpose of this project is to analyze the acoustic impedance and absorption properties of various flight approved materials currently and potentially used by NASA in its work with the International Space Station. These materials, consisting of Bisco, Acoustifoam, and other metallic foams, in addition to Durette, Kevlar, and other manufactured felts, will be used in an experimental procedure utilizing an impedance tube. This procedure uses two microphones at fixed positions from a material under test. An audio source, at the opposite end of the testing material, drives sound through the impedance tube and sweeps through a range of distinct frequencies. As the sound propagates down the tube, the two microphones measure the superposition of the driven, incident sound wave and the sound wave reflected off the test material. When used in conjunction with processing software packages, these microphone responses can be recorded and evaluated to produce complex impedance quantities as functions of frequency. By using these results as a means to measure sound absorption coefficients of specific materials, these tested, flight approved materials may be specifically arranged and utilized to both maximize efficiency and minimize excess noise. These possible applications will not only provide scientific data but also potentially affect astronauts on current and future missions for NASA.

3aEDb16. Design characterization and testing of a custom air horn. Frederick J. Ward and William Slaton (Physics & Astronomy Department, University of Central Arkansas, 201 Donaghey, Conway, AR 72034, wvslaton@uca.edu)

Construction and testing of an air horn can provide educational insight into how certain design decisions can influence resulting acoustic properties. Using readily available materials such as pvc pipe and tin sheeting, one can construct an air horn capable of producing sound waves in the 100+ decibel range and frequencies between 150 and 400 Hz. Upon completion of a prototype, many experimental opportunities are available. The degradation of sound intensity over a distance can be tested by use of a sound level meter. Due to the unidirectional behavior of the sound waves from the horn, samples from different distances and angles from the source can provide more understanding of how sound propagates as a wave in an open environment, as opposed to it being a simple directional wave. Upon completion of the

testing, changes to the initial construction design can be implemented to investigate the relationship between the new model's performance and the prototype's. The air horn provides many opportunities for experimentation and testing. For example, frequencies and sound intensity can be altered by making design adjustments such as: diaphragm size, diaphragm material, housing material, bell size, nozzle length, etc. With a better understanding of the inner workings of these sound sources, one could use this design as a blueprint to expand the concept to either much larger or much lower frequency ranges which have applications in many different fields of study.

3aEDb17. Objective and subjective differences between adjacent seats in a multipurpose hall. Michael J. Dick, Jenna M. Daly, and Michelle C. Vigeant (Acoustics Prog. and Lab., Dept. of Mech. Eng., University of Hartford, 200 Bloomfield Avenue, West Hartford, CT 06117, michelle.vigeant@gmail.com)

The purpose of this study was to evaluate differences between adjacent seats in terms of objective measures and subjective perception based on measurements and recordings taken in a mid-sized multipurpose hall. The work is based on the hypothesis that minimal differences should be found between nearby seats. Measurements were taken in three groups of nine adjacent seats. The differences between seats were analyzed in terms of the number of just noticeable differences (JNDs), using 1 dB for strength (G), 5% for early decay time (EDT) and 1 dB for clarity index (C80). The average differences between adjacent seats within a given group were approximately 2.5 JNDs for G, 4 JNDs for EDT and 2.5 JNDs for C80, which implies that these differences might be audible. However, these differences may be within measurement error. Differences in late lateral energy level (GLL), a measure of listener envelopment (LEV), were all less than 1 dB. A total of 35 test subjects were presented binaural recordings from two seat groups and were asked to evaluate LEV and acoustical quality. In general, subjects did not perceive differences between the recordings as hypothesized, with the exception of two cases where differences in LEV were statistically significant.

WEDNESDAY MORNING, 24 OCTOBER 2012 BASIE A1, 7:55 A.M. TO 10:00 A.M.

Session 3aMU

Musical Acoustics: Physics of the Blues

Andrew C. H. Morrison, Chair
Natural Science Dept., Joliet Junior College, Joliet, IL 60431

Chair's Introduction—7:55

Invited Papers

8:00

3aMU1. Physics of the blues—Scales, harmony, and the origin of blues piano styles. J. Murray Gibson (Northeastern University, 360 Huntington Ave, 115 Richards Hall, Boston, MA 02115, m.gibson@neu.edu)

The development of the equal-temperament scale was driven, not by compromise, but by the need to extend the composer's "palette" and increase the harmonic sophistication of western music. Many interesting musical idioms emerged from harmonic artifacts associated with equal temperament. In particular the blues "crushed-note" piano style resulted during the birth to the blues from the melding of this scale with the pure melodies and harmonies. In the presentation I will relate the history from a scientific perspective and illustrate with short keyboard performances. In homage to Kansas - the home of Scott Joplin -related ragtime piano idioms will also be covered. I'll conclude by emphasizing how music is an excellent medium that displays the intimate relationship between science and art, and is a great vehicle to teach science.

8:25

3aMU2. Sound radiated from vibrating trumpet bells and its share in the overall radiated sound of that instrument. Wilfried Kausel, Vasileios Chatziioannou (Inst. of Music Acoustics, Univ. of Music and Performing Arts, Anton-von-Webern-Platz 1, Vienna 1030, Austria, kausel@mdw.ac.at), and Thomas R. Moore (Department of Physics, Rollins College, Orlando, FL)

Dallas Blues, published in 1912 by Hart A. Wand, is often considered to be the first published blues song. In this work the trumpet is the lead voice, and this is still often the case in blues music. Indeed, it is difficult to overstate the importance of this instrument to the development of the genre. Recent research indicates that bell vibrations with circular symmetry, such as breathing modes and axial length oscillations (i.e., piston-like vibrations of the bell or rim), have the potential to affect the input impedance and pressure transfer function of the trumpet. This in turn can significantly affect the sound. Structural and acoustical finite element simulations of an axisymmetric trumpet bell show that the sound level radiated from the vibrating walls is of the same order of magnitude as the sound level radiated by the air column in a wide frequency band around the structural resonances. Since these axial structural resonances have a much wider bandwidth than elliptic modes, it is not necessary that air resonances exactly match structural resonances in order to produce a significant effect. The contribution of axial bell vibrations to the sound of the instrument is shown to be consistent with these simulations.

8:45

3aMU3. Guitar pickups—Where next? Mark French (MET, Purdue University, 138 Knoy Hall, 401 N. Grant St, West Lafayette, IN 47907, rmfrench@purdue.edu) and Davin Huston (Electrical Engineering Technology, Purdue University, West Lafayette, IN)

Electromagnetic guitar pickups have been in wide use for more than 50 years. Not only are the basic operational principles unmodified, some of the most popular pickups are exact copies of 50 year old designs. This situation begs the obvious question of where pickup design is headed and where there are opportunities for improvements. There are only a few underlying physical phenomena that can be harnessed to sense string motion. However, it would seem that there is plenty of room for innovative applications of familiar physical principles. This paper discusses current trends in pickup designs, suggests direction for further development and describes an effort to update the design of inductive pickups.

9:05

3aMU4. Pitch bending in the diatonic harmonica. James P. Cottingham (Physics, Coe College, 1220 First Avenue, Cedar Rapids, IA 52402, jcotting@coe.edu)

Pitch bending by wind instrument players is standard practice in many genres of music, but it is considered essential in blues harmonica playing. Some simple pitch bends involve the coupling of a single reed to a resonator such as the vocal tract of the player, but a full description of pitch bending in the harmonica involves consideration of the coupling between the two reeds, one for each direction of airflow, that share a single reed chamber. The most common pitch bends are those for which the primary reed in the reed chamber sounds a note higher in frequency than that sounded by the secondary reed. In these cases notes can be bent downward to pitches between those sounded by the two reeds. In addition, some players use more advanced techniques to bend notes beyond the frequencies of the chamber reeds. This paper reviews experimental work and theoretical modeling done on pitch bending in the harmonica during the last thirty years. This includes measurements made while pitch bends are produced by players as well as experiments on harmonicas in more conventional laboratory settings.

9:25

3aMU5. The harmonica as a blues instrument: Part I. Gordon Ramsey (Physics, Loyola University Chicago, 6460 N Kenmore, Chicago, IL 60626, gprspinphys@yahoo.com)

This is part one of two presentations on the same project. The modern harmonica, or harp, has been around since the early 19th century. It is typically used in blues, country, rock and roll and folk music. These musical genres are somewhat similar in structure and form, and often borrow ideas from each other. The harmonica is appropriate as a backup to the main vocal melody and instruments due to its rich harmonic structure and subdued intensity. The ability to apply vibrato and gradual slurs make it a perfect instrument to get the "bluesy" idea across. Our harp research group has investigated the physical properties of harmonica structure to illustrate how different structures lead to varied sounds, each of which is appropriate to a particular style of music.

Contributed Paper

9:45

3aMU6. The harmonica as a blues instrument: Part II. Joseph Wiseman, Chris Banaszak, and Gordon Ramsey (Physics, Loyola University of Chicago, Chicago, IL 60626, wsgywrest@gmail.com)

This is part two of two presentations on the same project. The modern harmonica, or harp, has been around since the early 19th century. It is typically used in blues, country, rock and roll and folk music. These musical genres are somewhat similar in structure and form, and often borrow ideas from each other. The harmonica is appropriate as a backup to the main vocal melody and instruments due to its rich harmonic structure and subdued intensity. The ability to apply vibrato and gradual slurs make it a perfect instrument to get the "bluesy" idea across. Our harp research group has investigated the physical properties of harmonica structure to illustrate how different structures lead to varied sounds, each of which is appropriate to a particular style of music.

Session 3aNS

Noise, Physical Acoustics, and Structural Acoustics and Vibration: Launch Vehicle Acoustics

Kent L. Gee, Cochair
Brigham Young University, Provo, UT 84602

R. Jeremy Kenny, Cochair
NASA, Huntsville, AL 35812

Invited Papers

8:00

3aNS1. Extension of a launch pad noise prediction model to multiple engines and directional receivers. Kenneth J. Plotkin (Wyle, 200 12th Street South, Arlington, VA 22202, kenneth.plotkin@wyle.com) and Bruce T. Vu (NASA, Kennedy Space Center, FL)

A model, PAD, has been developed for prediction of noise in the vicinity of launch vehicles, with original application to the mobile launcher and tower for the Ares I launch vehicle. It follows the basic principles of a traditional NASA model (NASA SP-8072, 1971), with updated source components, including impingement, water suppression and acoustic shielding by three dimensional launcher configurations. For application to Space Launch System, the model has been extended to multi-engine vehicles, using the plume merging model developed by Kandula, Vu and Lindsay (AIAA Paper 2005-3091) and accommodating multiple flame holes in the deck. The capability has also been added to account for receiver directivity. This can be significant when predicting the load on the surfaces of enclosures on the launch tower. It is also an issue for model scale tests (such as ASMAT) where microphones and their mounts are not small enough to be omnidirectional, and thus do not measure free field levels. (Work supported by the National Aeronautics and Space Administration.)

8:20

3aNS2. Full-scale rocket motor acoustic tests and comparisons with models: Revisiting the empirical curves. Michael M. James, Alexandria R. Salton (Blue Ridge Research and Consulting, LLC, 15 W. Walnut St., Suite C, Asheville, NC 28801, michael.james@blueridgeresearch.com), and Kent L. Gee (Department of Physics and Astronomy, Brigham Young University, Provo, UT)

Development of the next-generation space flight vehicles has prompted a renewed focus on rocket sound source characterization and near-field propagation modeling. Improved measurements of the noise near the rocket plume are critical for direct determination of the noise environment. They are also crucial in providing inputs to empirical models and in validating computational aeroacoustics models. NASA's SP 8072 acoustic load prediction model is a widely used method for predicting liftoff acoustics. SP-8072 implements two Distributed Source Methods (DSM-1 and DSM-2), which predict the loading as the sum of the radiated field from each source distributed along the plume. The prediction model depends largely on empirical curve fits computed from historical data to determine the source power and frequency content at distances along the plume. Preliminary results from measurements of a static horizontal firing of Alliant Techsystems Orion 50S XLG performed in Promontory, UT are analyzed with respect to the historical data that drives the SP-8072 prediction model.

8:40

3aNS3. Full-scale rocket motor acoustic tests and comparisons with models: Updates and comparisons with SP-8072. Michael M. James, Alexandria R. Salton (Blue Ridge Research and Consulting, LLC, 15 W. Walnut St., Suite C, Asheville, NC 28801, michael.james@blueridgeresearch.com), and Kent L. Gee (Department of Physics and Astronomy, Brigham Young University, Provo, UT)

Development of the next-generation space flight vehicles has prompted a renewed focus on rocket sound source characterization and near-field propagation modeling. Measurements taken during a static horizontal firing of Alliant Techsystems Orion 50S XLG are compared to the predicted levels produced from NASA's SP-8072 Distributed Source Methods (DSM-1 and DSM-2). Two modifications to the SP 8072 prediction model are considered in regards to the source directivity and the source power spectral distribution. All models considered provide a good first-order approximation given an appropriate total acoustic sound power. However, a more physical model is needed to adequately map out the critical near-field region as well as the far-field propagation. A new intensity-based measurement system and corresponding procedures are currently being developed for determining this near field energy flow and for achieving source characterization capabilities beyond traditional pressure measurements. These advances are believed to be a step toward improved measurements and modeling of the rocket plume.

9:00

3aNS4. Scale model acoustic test overview. Douglas Counter (NASA George C. Marshall Space Flight Center, Huntsville, AL) and Janice Houston (Jacobs ESTS Group, 1500 Perimeter Pkwy, Suite 400, Huntsville, AL 35806, janice.d.houston@nasa.gov)

Launch environments, such as lift-off acoustic (LOA) and ignition overpressure (IOP), are important design factors for any vehicle and are dependent upon the design of both the vehicle and the ground systems. LOA environments are used directly in the development of vehicle vibro-acoustic environments and IOP is used in the loads assessment. The Scale Model Acoustic Test (SMAT) program was implemented to verify the Space Launch Systems LOA and IOP environments for the vehicle and ground systems including the Mobile Launcher (ML) and tower. The SMAT is currently in the design and fabrication phase. The SMAT program is described in this presentation.

9:20

3aNS5. Frequency-based spatial correlation assessments of the Ares I subscale acoustic model test firings. Robert J. Kenny (NASA, Mail Stop ER42, Bldg 4203, Marshall Space Flight Center, Huntsville, AL 35812, robert.j.kenny@nasa.gov) and Janice Houston (Jacobs Engineering, Huntsville, AL)

The Marshall Space Flight Center has performed a series of test firings to simulate and understand the acoustic enviroments generated for the Ares I liftoff profiles. Part of the instrumentation package had special sensor groups to assess the acoustic field spatial correlation features for the various test configurations. The spatial correlation characteristics were evaluated for all of the test firings, inclusive of understanding the diffuse to propagating wave amplitude ratios, the acoustic wave decays, and the incident angle of propagating waves across the sensor groups. These parameters were evaluated across the measured frequency spectra and the associated uncertainties for each parameter were estimated.

9:40

3aNS6. Prediction of nonlinear propagation of noise from a solid rocket motor. Michael B. Muhlestein, Kent L. Gee, Derek C. Thomas, and Tracianne B. Neilsen (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602, mimuhle@gmail.com)

The extreme sound pressure levels radiated from rocket motors are such that nonlinear propagation effects can be significant. Here, free-field nonlinear propagation has been modeled for noise produced by a GEM-60 solid rocket motor. Measured waveforms were used as inputs into a numerical model based on the generalized Burgers equation. In both temporal and frequency domains the nonlinear predictions are significantly closer to the measured signals than free-field, linear predictions. In the temporal domain, shock coalescence and a transition from the weak-shock regime of propagation to the beginning of the old-age regime are clearly observed in both the nonlinear prediction and the measured data. These phenomena are completely missing in the linear prediction. In the frequency domain, continual transfer of energy upward in the spectrum reduces attenuation of high-frequency components when compared to predictions from the linear model. Various comparisons are made as a function of input distance for two different radiating angles from the rocket plume; these comparisons illustrate the importance of including nonlinear effects in rocket noise propagation modeling.

10:00–10:20 Break

10:20

3aNS7. Scale model tests for acoustic prediction and reduction of epsilon launch vehicle at lift-off. Seiji Tsutsumi (JEDI, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Tatsuya Ishii (APG, JAXA, Chofu, Tokyo, Japan), Kyoichi Ui, Sinichiro Tokudome (Epsilon Rocket Project Team, JAXA, Tsukuba, Ibaraki, Japan), and Kei Wada (Science Service, Inc., Chuuou-ku, Tokyo, Japan)

Test campaign using 1/42-scale model is conducted to predict acoustic level of the Epsilon launch vehicle at lift-off. Analogy between sub-scale and full-scale tests is investigated to obtain the same feature of the acoustics. Methodology to correct the measured data obtained in the sub-scale test for predicting the full-scale environment is also clarified in this study. The acoustic results around the practical shape of the launch-pad are successfully obtained. The parametric studies are conducted to reduce noise level in the test campaign with the help of the numerical simulation, and the effect for noise reduction is observed up to 5dB in 1/1-octaveband SPL.

10:40

3aNS8. Acoustic measurement of 1:42 scale booster and launch pad. Tatsuya Ishii, Seiji Tsutsumi, Kyoichi Ui, Shinichiro Tokudome (Japan Aerospace Exploration Agency, 7-44-1 Jindaijihigashi-machi Chofu-shi, Tokyo 182-8522, Japan, ishii.tatsuya@jaxa.jp), Yutaka Ishii (Bruel & Kjaer Japan, Tokyo, Japan), Kei Wada (Science Service, Inc., Tokyo, Japan), and Satoru Nakamura (Tokyo University of Science, Tokyo, Japan)

This paper describes the acoustic measurement of the subscale booster and launch pad. The 1:42 scale solid propellant booster was settled over the launch pad model. The launch pad model was designed to deflect the hot and high speed plume, aimed at mitigating the feedback Mach waves toward the vehicle. The launch pad plays a role in attenuating the sound due to the impingement of the plume and the deflector. To investigate the acoustic field with a different booster height, acoustic measurement was carried out. The measurement involved the conventional acoustic measurement and the sound source localization. The conventional measurement employed the near-field microphones around the booster model and the far-field microphones on an arc centered on the booster nozzle or the impingement point of the plume and the launch pad. In the sound source localization, a phased array microphone system was settled to focus the deflector exit. The obtained acoustic data helped revise the design of the launch pad model.

11:00

3aNS9. Analysis of noise from reusable solid rocket motor firings. Kent L. Gee (Dept. of Physics and Astronomy, Brigham Young University, N243 ESC, Provo, UT 84602, kentgee@byu.edu), R. Jeremy Kenny (NASA Marshall Space Flight Center, Huntsville, AL), Trevor W. Jerome, Tracianne B. Neilsen (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT), Christopher M. Hobbs (Wyle Laboratories, Arlington, VA), and Michael M. James (Blue Ridge Research and Consulting, Asheville, NC)

As part of investigations into the design of next-generation launch vehicles, near and far-field data were collected during horizontal static firings of reusable solid rocket motors. In addition to spectral analysis at individual microphone locations, the spatial variation of overall and one-third octave band pressure levels at sideline and polar arc arrays is considered. Analysis of the probability density functions reveals positively skewed pressure waveforms, but extreme skewness in the first-order estimate of the time derivative because of the presence of significant acoustic shocks.

Contributed Papers

11:20

3aNS10. Decomposition of military jet aircraft mixing noise into fine and large-scale turbulent components. Tracianne B. Neilsen, Kent L. Gee, Alan T. Wall (Dept. of Physics and Astronomy, Brigham Young University, N311 ESC, Provo, UT 84602, tbn@byu.edu), and Michael M. James (Blue Ridge Research and Consulting, Provo, Utah)

Many far-field measurements of laboratory-scale jet noise have shown good agreement with the two similarity spectra developed to represent the contributions of fine-scale and large-scale turbulent structures [Tam et al., AIAA paper 96-1716, 1996]. Measurements near an F-22A Raptor provide a means to study how accurately the similarity spectra describe the noise from a full-scale, high-performance, jet engine. Comparisons have been made using ground-based microphones at 60° to 150° for three engine conditions: intermediate, military and afterburner, with more detailed analyses than described previously [Neilsen et al., J. Acoust. Soc. Am. 129, 2242 (2011)]. The good agreement with Tam's predictions - the fine-scale spectrum at upstream and sideline angles and the large-scale spectrum in the maximum radiation direction - permits a quantitative analysis of the contributions of the two spectra at other angles. The radiated spectra and overall levels for all three engine conditions have been decomposed into contributions from the two spectra as functions of angle. Of particular interest is the appreciable contribution of fine-scale turbulence aft of the maximum radiation directions at military and afterburner conditions. [Work sponsored by the Office of Naval Research.]

11:35

3aNS11. Near-field correlation and coherence of military jet noise. Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen, Alan T. Wall (Dept. of Phys. and Astronomy, Brigham Young Univ., 562 N 200 E # 17, Provo, UT 84606, blaineharker@byu.net), Michael M. James (Blue Ridge Res. and Consulting, Asheville, Utah), and Sally A. McInerny (Dept. of Mech. Eng., Univ. of Louisiana, Lafayette, LA)

Correlation and coherence analyses in the near field of military jet noise provide insight into source and radiation characteristics. Data were measured parallel to the exhaust centerline of an F-22A Raptor and spatial correlation and coherence values were calculated. Low spatial correlation at the sideline indicates radiation dominated by multiple incoherent sources. In contrast, the downstream region is characterized by high spatial correlation, suggesting radiation primarily from large-scale turbulent structures. Variations in spatial correlation in the axial direction can be related to the spectral dependence on measurement location, which supports the idea of a two-source jet-noise model. Coherence calculations, which decompose the correlation information into narrow frequency bands, further support this idea. [Work supported by the Office of Naval Research.]

WEDNESDAY MORNING, 24 OCTOBER 2012 LESTER YOUNG A, 8:00 A.M. TO 11:45 A.M.

Session 3aPA

Physical Acoustics: Thermoacoustics, Physics, and More

Josh R. Gladden, Chair
Physics & NCPA, University of Mississippi, University, MS 38677

Contributed Papers

8:00

3aPA1. Dynamic stabilization of the Rayleigh-Bénard instability in a cubic cavity. Randy M. Carbo (Graduate Program in Acoustics, Penn State University, State College, PA), Robert W. Smith, Matthew E. Poese (Applied Research Laboratory, Penn State University, P.O. Box 30, State College, PA 16804, rws100@psu.edu), and Anand Swaminathan (Graduate Program in Acoustics, Penn State University, State College, PA)

The dynamic stability of the Rayleigh-Bénard instability with vertical vibration in a cubic container is computationally modeled. Two periodic parametric drives are considered (sinusoidal and rectangular), as well as two thermal boundary conditions on the sidewalls (insulating and conducting). The linearized equations are solved using a spectral Galerkin method and Floquet analysis. Floquet analysis recovers both the synchronous and the subharmonic regions of instability. The conditions necessary for dynamic stability are reported for Rayleigh numbers from critical to 10^7 and for Prandtl numbers in the range of 0.1-7.0, and the approach produces maps over a wide range of Rayleigh number and vibration parameters for stability. The linear model is compared to data set available in the literature [G. W. Swift and S. Backhaus J. Acoust. Soc. Am. **126**, 2273 (2009)] where the performance of system simulating an inverted pulse tube cryocooler is measured. The relevant instability for this case is the synchronous instability. Over this limited data set, the model appears to bound the empirically observed conditions for stability, but in some cases the model would seem to predict significantly higher required periodic acceleration amplitudes that appear to have been observed by Swift. Comparison with another data set is on-going. [Research supported by the Office of Naval Research and ARL Exploratory and Foundational Research Program.]

8:15

3aPA2. Thermoacoustic device for nuclear fuel monitoring and heat transfer enhancement. Randall A. Ali, Steven L. Garrett (Grad. Prog. in Acoustics, Penn State University, Grad. Prog. in Acoustics, P.O. Box 30, State College, PA 16804, randallali@gmail.com), James A. Smith, and Dale K. Kotter (Fundamental Fuel Properties Group, Idaho National Laboratory, Idaho Falls, ID)

The Fukushima Dai'ichi nuclear disaster of 2011 exposed the need for self-powered sensors that could transmit the status of the fuel rods within the reactor and in spent fuel ponds that was not dependent upon availability of external electrical power for either sensing or telemetry. One possible solution is the use of a thermoacoustic standing wave engine, incorporated within a fuel rod, which is heated by the nuclear fuel. The engine's resonance frequency is correlated to the fuel rod temperature and will be transmitted by sound radiation through the reactor's or storage pond's surrounding water. In addition to acting as a passive temperature sensor, the

thermoacoustic device will serve to enhance heat transfer from the fuel to the surrounding heat transfer fluid. When activated, the acoustically-driven streaming flow of the gas within the fuel rod will circulate gas away from the nuclear fuel and convectively enhance heat transfer to the surrounding coolant. We will present results for a thermoacoustic resonator built into a Nitonic® 60 (stainless steel) fuel rod that can be substituted for conventional fuel rods in the Idaho National Laboratory's Advanced Test Reactor. This laboratory version is heated electrically. [Work supported by the U.S. Department of Energy.]

8:30

3aPA3. Addition of a series spring to a lumped element model of a novel thermoacoustic refrigerator. Eric C. Mitchell, Steven L. Garrett, Robert W. M. Smith, Matthew E. Poese, and Robert M. Keolian (Applied Research Laboratory, Penn State University, State College, PA 16804, emitchell756@gmail.com)

A lumped element model is introduced for an "in-line" traveling-wave thermoacoustic refrigerator, based on a recent patent by Backhaus and Keolian [US Pat. No. 7,908,856 (22 Mar 2011)]. This model couples three electro-mechanical motors to the acoustical domain using a flexure sealed piston. Results of this lumped-element model were found to be in good agreement with a DELTAEC model. Since large displacements result in flexure fatigue, the addition of a series spring between the motor and piston was evaluated to reduce flexure seal displacement while matching the optimum load impedance. The assignment of the correct dynamic mass of the series spring when both the motor and flexure ends of the spring are moving has not been addressed previously in the literature. To determine the two effective masses required at each end of the spring, the spring is discretized into a large number of elements and is converted into a simplified circuit model to assign values to lumped-element components. These values depend upon an initial knowledge of both end velocity magnitudes and phases. This enables an iterative solution for the effective spring masses since velocities need to be known for effective masses to be determined and vice versa. [Work supported by the Applied Research Laboratory and the U.S. Department of Energy.]

8:45

3aPA4. Nusselt numbers of laminar, oscillating flows in stacks and regenerators with pores of arbitrary cross-sectional geometry. John Brady (Los Alamos National Laboratory, MSD429, Los Alamos, NM 87544, jbrady@lanl.gov)

General expressions for the Nusselt numbers of laminar oscillating flows within the pores of stacks and regenerators are derived from thermoacoustic theory developed by Rott and Swift. These expressions are based on bulk (velocity-weighted, cross-sectionally averaged) temperature, rather than the cross-sectionally averaged temperature. Two cases are considered: flow with oscillating pressure and no external temperature gradient, and oscillating velocity within an external temperature gradient and negligible pressure oscillations. These expressions are then applied to parallel plates, circular pores, rectangular pores, and within the boundary layer limit. Steady-flow Nusselt numbers are recovered when the thermal penetration depth is at least as great as the hydraulic radius of the pore. In addition, temperature and flow profiles within this regime are like those of steady flows.

9:00

3aPA5. Complex intensity in circular ducts containing an obstruction. Ray Kirby (Mechanical Engineering, Brunel University, Uxbridge, Middlesex UB8 3PH, United Kingdom, ray.kirby@brunel.ac.uk), Jevgenija Prisutova (School of Engineering, University of Bradford, Bradford, West Yorkshire, United Kingdom), Wenbo Duan (Mechanical Engineering, Brunel University, Uxbridge, Middlesex, United Kingdom), and Kirill Horoshenkov (School of Engineering, University of Bradford, Bradford, West Yorkshire, United Kingdom)

Sound intensity may be defined as a complex quantity in which the real part of the intensity is related to the magnitude of the local mean energy flow, and the imaginary part to the local oscillatory transport of energy. By treating intensity as a complex quantity it is possible to visualise energy flow in a different way and this has the potential to aid in the interpretation of, say, sound fields scattered by objects. Accordingly, the sound field scattered by an object placed in a semi-infinite circular duct is examined here. Experimental measurements of complex intensity are obtained in three (orthogonal) directions using a Microflown intensity probe, and measurements are compared to predictions obtained using a finite element based theoretical model. Comparisons between prediction and measurement are undertaken for both plane wave and multi-modal sound fields and here it is noted that when at least one higher order mode propagates it becomes more difficult to obtain good agreement between prediction and experiment for the complex intensity.

9:15

3aPA6. Modeling interferometric sensor response to the photoacoustic effect in layered systems. Logan Marcus, Richard Raspet, and Vyacheslav Aranchuk (NCPA, University of Mississippi, 1 Coliseum Drive, NCPA Room 1101, University, MS 38677, lsmarcus@olemiss.edu)

Chemical elements and molecules have characteristic absorption spectra that can be used for identification and detection. The photoacoustic effect has previously been used to perform spectroscopic measurements. We describe a modified photoacoustic spectroscopy method to accomplish standoff detection of thin layers of materials using an interferometric sensor. The interferometric sensor measures changes to the optical path length of an interrogation beam incident on a surface. We have developed a detailed model of the physical processes that result when a system comprised of a thin layer on a larger substrate is excited by the absorption of a modulated Gaussian laser beam. The modulated excitation beam generates heating in the sample which leads to surface motion, modulation of the temperature profile in the adjacent air, and an acoustic wave which all contribute to the signal. The model allows for the calculation of the measured signal of the interferometric sensor using the physical properties of the sample and the excitation beam. The presented model and experimental work are all applied to an idealized system comprised of a thin layer of gold on a low absorptivity borosilicate substrate to validate the computation. Future work will extend to a variety of layers and substrates.

9:30

3aPA7. Acoustic radiation force and radiation torque on Rayleigh particles. Tiago P. Lobo and Glauber T. Silva (Physical Acoustics Group - IF, UFAL, Instituto de Física, Campus A. C. Simões - Av. Lourival Melo Mota, s/n, Cidade Universitária, Maceió, Alagoas 57072-900, Brazil, tomaz.glauber@gmail.com)

In this work, the acoustic radiation force and radiation torque exerted by an arbitrary shaped wave on a spherical particle in the Rayleigh approximation (i.e. the incident wavelength is much smaller than the particle dimensions) are discussed. The host fluid in which the particle is suspended is assumed to be inviscid. Expressions for the force and the torque are obtained in terms of the incident acoustic fields, namely pressure and particle velocity. As it turns out, the obtained radiation force expression represents a generalization of Gor'kov's formula [Sov. Phys. Dokl. 6, 773-775 (1962)]. Moreover, the radiation torque can be expressed in terms of the incident Reynolds' stress tensor. The method is applied to calculate both radiation force and radiation torque produced by Bessel beams. It is demonstrated that only the first-order Bessel vortex beam generates radiation torque on a particle placed in the beam's axis. In addition, results involving off-axial particles and Bessel beams of different order are illustrated.

9:45

3aPA8. Modeling of photoacoustic Raman spectroscopy with dissipation. David Chambers and Chance Carter (Lawrence Livermore National Laboratory, PO Box 808, Livermore, CA 94551, chambers2@llnl.gov)

Photoacoustic Raman spectroscopy (PARS) is a technique used to identify chemical species mixed in a gas or liquid based on their pattern of vibrational energy levels. Raman spectroscopy differs from the more familiar absorption spectroscopy by using a nonlinear two-photon process that can be more sensitive to small differences in vibrational energy levels. Thus it can detect defect sites in solid-state optical materials, or low concentrations of chemical species in gases. The Raman scattering process generates acoustic pulses that can be detected with a microphone. In this talk we present an

overview of PARS and present an updated model that includes dissipation for the production of acoustic pulses from the energy deposited in the medium during the Raman scattering process. We also show some preliminary measurements of the process for Raman conversion in hydrogen. This work performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract DE-AC52-07NA27344.

10:00–10:15 Break

10:15

3aPA9. Impact of rigid sphere scattering on measurement of acoustic shocks. Michael B. Muhlestein, Derek C. Thomas, and Kent L. Gee (Department of Physics and Astronomy, Brigham Young University, Provo, UT 84602, mimuhle@gmail.com)

Multi-microphone arrays embedded in a rigid spherical housing have been used to estimate field quantities such as vector intensity. However, the measured pressure waveforms are modified by the scattering of the incident pressure wave on the sphere. The frequency response function and the corresponding impulse response function for microphones in a sphere can be created using Mie scattering theory. This permits analysis of the scattering effect on pressure measurements for shock-containing broadband waveforms, such as those produced by solid rocket motors. The abrupt pressure rises associated with shocks are seen to be either overestimated or underestimated depending on the angle of the measuring microphone. These shocks are the most affected portion of the waveform due to their high frequency content. In addition, deconvolution of measured rocket signals using the predicted impulse responses of the microphones in the array provides an apparently consistent estimate of the free-field signal at the probe center.

10:30

3aPA10. Actuating micro devices with acoustic white noise. Raul Esquivel-Sirvent (Instituto de Fisica, UNAM, Apdo Postal 20-364, Mexico DF 01000, Mexico, raul@fisica.unam.mx)

We present a theoretical calculation of the actuation of a model micro system, such as a microelectromechanical system (MEMS), by the acoustic pressure of white noise. This is a classical analog of the Casimir effect, thus the name Casimir acoustic pressure. Unlike the quantum case, the acoustic Casimir pressure can be attractive or repulsive depending on the frequency bandwidth of the acoustic noise. As a case study, a one-degree-of-freedom simple-lumped system in an acoustic resonant cavity is considered. By properly selecting the frequency bandwidth of the acoustic field, the acoustic pressure can be tuned to increase the stability in existing microswitch systems by selectively changing the sign of the force. The acoustic intensity and frequency bandwidth are introduced as two additional control parameters in capacitive micorswitches. Applications of this concept in microfluidics will be also discussed.

10:45

3aPA11. Infrasound scattering by the Lamb dipole vortex. Konstantin Naugolnykh (NOAA/Zeltech, 325 Broadway, Boulder, CO 80305, konstantin.naugolnykh@noaa.gov)

The infrasound scattering by the Lamb dipole is considered in the present paper in Born approximation and using the asymptotic presentation for the Green function of the scattered field. The Lamb dipole consists of two vortexes rotating in the opposite direction what induces the specific features of process of scattering infrasound by this object. They probably accompanied the infrasound generation by high-intensity atmospheric events such as cyclones.

11:00

3aPA12. The nonlinearity parameter, B/A, in FC-43 Fluorinert up to 373 K and 13.8 MPa. Blake T. Sturtevant, Cristian Pantea, and Dipen N. Sinha (Materials Physics and Applications, Los Alamos National Laboratory, PO Box 1660, Los Alamos, NM 87545, bsturtev@lanl.gov)

Acoustic imaging systems based on the parametric array concept utilize a nonlinear medium for mixing high frequency sound into a beam of low frequency collimated sound. Fluorocarbon fluids are very appealing as nonlinear mixing media with values of the nonlinearity parameter, B/A, typically greater than 10 at ambient temperature and pressure. To design acoustic imaging systems for high temperature and high pressure environments, such as found in geothermal and petroleum wells, it is important to know how B/A varies with temperature and pressure. This work reports the determination of B/A in FC-43 at temperatures up to 373 K and pressures up to 13.8 MPa using the thermodynamic method. Sound velocities were measured using Swept Frequency Acoustic Interferometry at 11 pressures between ambient and 13.8 MPa along 6 isotherms between ambient and 373 K. A 3rd order least-squares fit of measured sound speeds was used to determine temperature and pressure dependence. The B/A of FC-43 was found to increase with both temperature and pressure near ambient conditions and to go through a maximum around 340 K and 6 MPa.

11:15

3aPA13. Acoustic streaming in channel bifurcated by an elastic partition. Megha Sunny, Taoufik Nabat, and Charles Thompson (Electrical and Computer Engineering, University of Massachusetts Lowell, 1 University Ave, Lowell, MA 01854, taoufik_nabat@student.uml.edu)

Fluid motion in a narrow channel partitioned along its long axis by a flexible membrane is examined. The enclosed fluid is excited by the time harmonic displacement in the channel cross-section. The acoustic streaming that ensues and its impact of microfluidic transport is of particular interest in this work. A three-dimensional regularized Stokeslet based analysis is presented for fluid motion in the low streaming Reynolds number regime. The effect of frequency modulation on fluid motion is examined.

11:30

3aPA14. Acoustic scattering from dual frequency incident fields. Chrisna Nguon, Max Denis, Kavitha Chandra, and Charles Thompson (Electrical and Computer, University of Massachusetts Lowell, Lowell, MA 01854, chrisna_Nguon@student.uml.edu)

The pressure field produced by the spatial interaction of two high frequency incident beams in a three-dimensional scattering object is investigated. Of particular interest is the radiated pressure produced in response to the difference-frequency component generated from the non-linear interaction between the beams and the scattering medium. The influence of high acoustic contrast and resonant scattering is considered in the analysis. This work presents a computational study of the scattered pressure that results from the Reynolds stress in a fluid scatterer. Using Padé approximants, it is shown that the stress tensor can be computed using a uniform expansion in the contrast gauge for the scattered pressure. This allows one to investigate scattering volumes characterized by high compressibility contrast.

Session 3aPP

Psychological and Physiological Acoustics: Perception and Models

G. Christopher Stecker, Chair
Speech and Hearing Sciences, University of Washington, Seattle, WA 98105

Contributed Papers

8:30

3aPP1. Infants' ability to separate superimposed vowels. Lynne Werner, Bonnie Lau, and Ashley Flad (Speech & Hearing Sciences, University of Washington, 1417 North East 42nd Street, Seattle, WA 98105-6246, lawerner@u.washington.edu)

Three- and seven-month-old infants were tested using an observer-based procedure in three tasks to assess sound source segregation and selective attention. The stimuli were tokens of the vowels /a/ and /i/, spoken by two male talkers, 519 ms in duration, presented at 70 dB SPL. Success was defined as achieving 80% correct in fewer than 40 test trials. In the first task, infants heard one vowel spoken by one talker repeated at 1319 ms intervals. They learned to respond when the talker changed on one repetition of the vowel. In the second task, the tokens of the two talkers producing the same vowel were superimposed. Infants heard combined tokens repeatedly and learned to respond when the vowel produced by one talker changed. In the third task, either talker could produce the changed vowel. Infants learned to respond when one talker, but not the other, produced the changed vowel. Nearly all infants succeeded in the first two tasks. Nearly all 7-month-olds, but few 3-month-olds succeeded at the third task. These results suggest that the ability to selectively attend to one of two easily discriminable voices matures after the ability to segregate those voices. [Work supported by R01DC00396 and P30DC04661.]

8:45

3aPP2. Off-frequency masking effects on intensity discrimination. Harisadhan Patra (Audiology & Speech Pathology, Bloomsburg University, 226 CEH, 400 E 2nd Street, Bloomsburg, PA 17815, hpatra@bloomu.edu), Scott Seeman (Department of Communication Sciences & Disorders, Illinois State University, Normal, IL), Adam Burkland, Joseph Motzko, and Erin Lolley (Audiology & Speech Pathology, Bloomsburg University, Bloomsburg, PA)

Intensity discrimination, where a listener detects an intensity increment in an equal duration sinusoid or pedestal, is often used as a measure of intensity resolution. Intensity discrimination may be considered as tone-in-tone masking, where the pedestal is the masker and the increment is the signal. Despite the similarity between intensity discrimination and tone-in-noise masking, research suggests that a high-pass noise outside the critical band centered on the signal frequency adversely affects listeners' intensity-discrimination thresholds. The present study examines the limits of off-frequency masking effects on intensity discrimination in five normal-hearing young adults. Detection thresholds for a 50-ms increment, added to a 50-ms-long 1000-Hz pedestal in phase, were obtained in quiet and notched-noise (NN) conditions. The pedestal and noise levels were 60 dB SPL. NN stimuli were generated by filtering telegraph noise. The low-frequency cutoffs of the NN-notches were 188, 250, 375, 500, and 750 Hz while the high-frequency cutoffs were 1500, 2000, 3000, 4000, and 6000 Hz. The detection thresholds were poorer in NN conditions than in quiet, even when cutoff frequencies were more than one octave away from the signal frequency. Effects of off-frequency maskers on the psychometric functions are discussed. [Supported by BU research and scholarship grant.]

9:00

3aPP3. Perceptual weights for loudness reflect central spectral processing. Suyash N. Joshi and Walt Jesteadt (Psychoacoustics Laboratory, Boys Town National Research Hospital, 555 N 30th St., Omaha, NE 68131, Suyash.Joshi@boystown.org)

Weighting patterns for loudness obtained using the reverse correlation method are thought to reveal the relative contributions of different frequency regions to total loudness, the equivalent of specific loudness. Current models of loudness assume that specific loudness is determined by peripheral processes such as compression and masking. Here we test this hypothesis using 20-tone harmonic complexes (200Hz f0, 200 to 4000Hz, 250 ms, 65 dB/Component) added in opposite phase relationships (Schroeder positive and negative). Due to the varying degree of envelope modulations, these time-reversed harmonic complexes have been shown to produce different outputs at the basilar membrane and different amounts of forward and simultaneous masking. The perceptual weights for loudness did not differ for these two complexes. To determine whether the level rove introduced to obtain weights had changed the fundamental differences in the stimuli, a similar level rove ($\pm$8 dB) was introduced on each component of Schroeder positive and negative forward maskers. The Schroder negative maskers continued to be more effective. These results suggest that perceptual weights for loudness are not completely determined by peripheral processes and reflect a central frequency weighting template. [Work supported by NIH R01 DC011806 and P30 DC004662.]

9:15

3aPP4. Temporal weighting of interaural time and level differences carried by broadband noises. G. C. Stecker (Speech and Hearing Sciences, University of Washington, 1417 NE 42nd St, Seattle, WA 98105, cstecker@uw.edu)

Localization of real sounds involves integrating acoustic spatial cues as they evolve over time. This study measured binaural sensitivity over time, in the form of temporal weighting functions (TWFs) for trains of of noise bursts. Each stimulus comprised sixteen 1-ms bursts of white noise, presented at an interval (ICI) of 2 or 5 ms. In separate conditions, noise samples were either repeated ("frozen") or newly generated ("fresh") across bursts. On each of many trials, listeners indicated the apparent lateral position of a stimulus along a horizontal scale displayed on a touch-sensitive device. Lateral positions varied across trials as interaural time (ITD) and level (ILD) differences ranged +/-500 μs ITD or +/-5 dB ILD. Interaural differences of individual bursts in each train received additional random variation (ranging +/-100 μs and +/-2 dB) to allow calculation of TWFs by multiple linear regression of normalized responses onto per-burst ITD and ILD values. Consistent with past studies, TWFs for "frozen" noise-burst trains demonstrated large ICI-dependent weights on the initial burst ("onset dominance"), elevated weights near offset, and lower weights for interior bursts. Flatter TWFs, smaller onset/offset weights, and greater interior weights were measured for "fresh" vs "frozen" noise burst trains. [Supported by R01 DC011548.]

9:30

3aPP5. The relative contribution of dynamic and spectral cues in virtual sound source localization. Chengyun Zhang and Bosun Xie (Physics Department, School of Science, South China University of Technology, Wushan Rd. 381., Tianhe District, Guangzhou 510640, China, phbsxie@scut.edu.cn)

It is well known that the dynamic cues caused by head turning and pinna-based spectral cue are vital to sound source localization, especially for front-back and vertical localization. A series of localization experiment is carried out via virtual auditory display to explore the relative contribution of dynamic and spectral cues to localization. Virtual sources at different intended spatial locations are recreated using various combinations with noise stimuli at full audio or 4 kHz-lowpass frequency ranges, individualized HRTFs or non-individual HRTFs derived from KEMAR and spherical-head model, and static or dynamic binaural synthesis. Furthermore, statistical analyses are performed on localization performance in terms of the percentage of front-back and up-down confusion as well as the mean angle error in virtual source localization. Primary results indicate that both dynamic and spectral cues contribute to front-back and vertical localization; whereas dynamic cue contributes more to front-back localization, while individualized spectral cue at high frequency above 4 kHz contributes more to vertical localization. Dynamic and high-frequency individualized spectral cues are also helpful to reduce the angle error in localization. [Work supported by the National Natural Science Foundation of China,11174087, 50938003, and State Key Lab of Subtropical Building Science, South China University of Technology.]

9:45

3aPP6. Approximately calculate individual near-field head-related transfer function using an ellipsoidal head and pinnae model. Yuanqing Rui, Guangzheng Yu, and Bosun Xie (Physics Department, School of Science, South China University of Technology, Wushan Rd. 381#, Tianhe District, Guangzhou, Guangdong 510640, China, scgzyu@scut.edu.cn)

Head-related transfer functions (HRTFs) describe the acoustic transmission from a point source to ears and are individual-dependent. Measurement is an effective way to obtain individual HRTFs, but the workload is very heavy. This is particularly difficult in the near-field HRTFs measurement due to their source distance-dependence within 1.0 m. Numerical calculation is another way to obtain HRTFs. To reduce the calculation load, an ellipsoid head and pinnae (HAP) model is proposed in the present work for approximately calculating the individual near-field HRTFs via the fast multipole boundary method (FMBEM). The dimension of ellipsoid head is defined by head width, head height, head depth, which is selected to approximately fit the individual interaural time difference (ITD) and interaural level difference (ILD) below about 5 kHz. The individual pinna geometry obtained via a 3D laser scanner provides individual HRTF spectral cues above 5 kHz. To validate the proposed method, the HAP of KEMAR and human is constructed, and calculating results are compared with the measurements. Psychoacoustic experiment is also carried out to evaluate the model and results. [Work supported by the National Natural Science Foundation of China for Young Scholars (No 11104082) and the Natural Science Foundation of Guangdong Province.

10:00–10:15 Break

10:15

3aPP7. Prior probabilities tune attentional bandwidth. Michael Wolmetz and Mounya Elhilali (Center for Language and Speech Processing, Johns Hopkins University, 3400 N Charles St, Barton Hall, Baltimore, MD 21218, mikew@jhu.edu)

Top-down schemas or prior probabilities associated with different auditory objects and their acoustic features are thought to improve auditory scene analysis and auditory object recognition processes. This study focused on whether listeners implicitly track the prior probabilities associated with different tone frequencies, and whether tracking those probabilities modulates attentional bandwidth to improve detection sensitivity. Using the target-probe paradigm, attentional bandwidth for varying levels of probe probability was estimated for 40 listeners. To estimate the attentional bandwidth, probes were presented in a continuous band-pass noise masker at distances of .75, 1.25, 1.75, and 3.75 Equivalent Rectangular Bandwidths from a target frequency of 250 Hz, at varying probabilities of occurrence. All tones were presented at approximately 90% detection when presented alone as measured during a separate thresholding procedure. Results indicate that prior probability plays a role in attentional bandwidth: the attentional band is more broadly tuned when tones at probe frequencies are more likely, consistent with optimal observer predictions. Sensitivity of attentional bandwidth to endogenous (listener-directed) and exogenous (stimulus-driven) attention will be discussed.

10:30

3aPP8. What is a good musical pattern? On acoustic structure and goodness of pattern. Ronaldo Vigo (Psychology, Ohio University, 211 Porter Hall, Athens, OH 45701, vigo@ohio.edu), Yu Zhang (Communication Sciences and Disorders, Ohio University, Athens, OH), and Mikayla Barcus (Psychology, Ohio University, Athens, OH)

An open problem in acoustical psychophysics involves the extent to which the structuro-acoustical properties of a set of sounds determine goodness of pattern judgments (i.e., how good a pattern is perceived to be). We investigated this question experimentally and theoretically from the standpoint of the dimensional structure of sets of sounds defined over the dimensions of timbre, tone, and duration. We observed a distinctive goodness of pattern ordering for the structures tested which involved sets of four tones defined over the three dimensions. The results were consistent with predictions from categorical invariance theory (CIT; Vigo, 2009, 2011a, 2012) which posits that humans detect the dimensional invariants of a set of discrete sounds — a process described in the theory as structural kernel extraction. Using CIT to interpret our results, we also found that sets of stimuli in "structural equilibrium" (the condition when all of the dimensions of a stimulus set play the same structural role in the set) were perceived as conveying better sound patterns. Based on these results, we propose that the tradeoff between complexity and invariance described in CIT can account for differences in music preferences and style.

10:45

3aPP9. Efficient coding of multiple nonorthogonal redundancies between acoustic dimensions in novel complex sounds. Christian Stilp (Department of Psychological and Brain Sciences, University of Louisville, Life Sciences Building, Louisville, KY 40292, christian.stilp@gmail.com) and Keith Kluender (Speech, Language, and Hearing Sciences, Purdue University, West Lafayette, IN)

Stilp and colleagues (*Proc. Natl. Acad. Sci*. [2010]; *JASA* [2011]; *PLoS One* [2012]) provided perceptual evidence for efficient coding of robust covariance between acoustic dimensions in novel complex sounds. Discrimination of sounds that directly violated this redundancy (Orthogonal condition) was initially inferior to that of sounds obeying overall redundancy (Consistent condition). Performance was consistently predicted by principal components analysis (PCA), as experimental conditions aligned with statistical dimensions in PCA. Stilp and colleagues suggested efficient coding may contribute to perceptual organization for speech, but two aspects of their experimental designs qualify this extension: robust evidence for only one statistical regularity between acoustic dimensions was tested while speech possesses many; and, all statistical structure was mutually orthogonal which is often not true of speech sounds. Here, listeners discriminated sounds supporting two concurrent, nonorthogonal regularities (patterns of covariance between acoustic dimensions: attack/decay and spectral shape.) Despite nonorthogonality, these concurrent statistical regularities were efficiently coded, as discrimination of Consistent sound pairs was initially superior to that of Orthogonal sound pairs. Performance did not adhere to predictions made by PCA. Implications for speech perception and auditory 'category' acquisition will be discussed. [Supported by NIDCD.]

11:00

3aPP10. Modeling normal and hearing-impaired monaural and binaural signal detection in the presence of noise. Miriam Furst (Tel Aviv University, Ramat Aviv, Tel Aviv 69978, Israel, mira@eng.tau.ac.il)

Psychoacoustical investigations have demonstrated that monaural and binaural signal detection is affected by existence of distracting noise. The most prominent phenomena are co-modulation masking release (CMR) in monotic listening, and binaural masking level difference (BMLD) in diotic listening. Both CMR and BMLD are significantly deteriorates in hearing-impaired listeners. Both CMR and BMLD phenomena are tested by a complete model of the auditory system for normal and hearing-impaired systems. Prediction of the amplitude discrimination results are obtained by deriving the Cramer Rao lower bound (CRLB) of the neural activity. The auditory system model includes a complete cochlear model with integrated outer hair cells and tectorial membrane; an inner hair cell-synapse model that transduce the cilia motion to auditory nerve instantaneous rate; Inferior Colliculus that receives inputs from both ears and process them by excitatory-inhibitory (EI) cells. The AN activity is considered as a non-homogeneous Poisson process (NHHP). We have recently showed that EI cells are NHHP as well, if their inputs behave as NHHP. Therefore, CRLB can be derived analytically from both AN and IC outputs. We have successfully predicted major CMR and BMLD properties as a function of frequency and noise properties for normal and impaired auditory system.

11:15

3aPP11. Response to a pure tone in a nonlinear frequency-domain model of the cochlea. Julien Meaud and Karl Grosh (Mechanical Engineering, University of Michigan, Ann Arbor, MI 48104, jmeaud@Umich.edu)

The nonlinear response of the cochlea to a pure tone is simulated using a novel computational model. In this physiologically-based finite element model, the three-dimensional intracochlear fluid dynamics are coupled to a micromechanical model of the organ of Corti and to electrical potentials in the cochlear ducts and outer hair cells (OHC). Active feedback due to OHC somatic electromotility is represented by linearized piezoelectric relations and is coupled to the nonlinear hair-bundle mechanoelectrical transduction current. Using an alternating frequency/time method and a single set of parameters, we simulate the compressive nonlinearity, harmonic distortion and DC shifts in the response of the cochlea to a single tone. Model predictions agree well with available experimental data.

11:30

3aPP12. Pole-zero characterization of middle ear acoustic reflectance data. Sarah Robinson, Cac Nguyen, and Jont Allen (Electrical Engineering, University of Illinois at Urbana-Champaign, Urbana, IL 61801, srrobin2@illinois.edu)

Middle ear acoustic reflectance (AR) measurements have valuable clinical applications. AR is measured using a foam-tipped probe sealed in the ear canal, containing a microphone and receiver (i.e. MEPA3 system, Mimosa Acoustics). From the complex pressure response to a broadband stimulus, the acoustic impedance and reflectance of the middle ear can be calculated as functions of frequency. A sizeable pool of normal and pathological AR data, collected by various researchers, indicates that many pathological ears have an AR that systematically differs from normative data. Assessment of such data typically relies on consideration of the magnitude AR, or separate consideration of AR magnitude and phase. By fitting poles and zeros to AR data, we have achieved a compact and accurate representation of the complex data (<5% RMS relative error). It was found that removing an approximated ear canal phase delay from AR data before fitting allowed for better comparison across ears, and better comparison with existing network models of the middle ear. Pole-zero fits indicated isolated regions of individual variation for normal ears, and showed differences between normal and pathological ears. Pole-zero fitting shows promise for more quantitative, robust diagnosis of middle ear pathology than AR magnitude alone.

WEDNESDAY MORNING, 24 OCTOBER 2012 — LIDO, 10:15 A.M. TO 12:00 NOON

Session 3aSA

Structural Acoustics and Vibration: Structural Acoustics Optimization

Micah R. Shepherd, Chair
Applied Research Lab, Penn State University, State College, PA 16801

Invited Papers

10:15

3aSA1. Structural acoustic optimization of ribbed panels excited by complex forcing functions. Micah R. Shepherd and Stephen A. Hambric (Applied Research Lab, Penn State University, PO Box 30, State College, PA 16801, mrs30@psu.edu)

Structural acoustic optimization is performed on a ribbed panel exited by diffuse acoustic field and turbulent boundary layer (TBL) flow. In order to vary the rib location during the optimization, component mode synthesis (CMS) was used with the rib and plate treated as separate substructures. The CMS approach couples the individual modes of the rib and plate using impedances at the connection points thus only requiring full eigenanalysis and radiation resistance calculations once and allowing for more rapid function evaluation during the optimization. The optimization was then performed using an evolutionary strategy with covariance matrix adaption. The rib location and properties were varied by the optimizer to find the best low-noise panel. An exhaustive search was performed to verify the optimum solution and compare results for several objective functions. Alternative design variables and constraints will also be discussed.

10:35

3aSA2. Sound reduction from vibrating thin plates using dimpling and beading design. Kyle R. Myers (Mechanical and Aeronautical Engineering, Western Michigan University, Kalamazoo, MI), Nabeel T. Alshabatat (Mechanical Engineering, Tafila Technical University, Tafila, Jordan), and Koorosh Naghshineh (Mechanical and Aeronautical Engineering, Western Michigan University, 1903 W. Michigan Ave., M/S 5343, Kalamazoo, MI 49008, koorosh.naghshineh@wmich.edu)

This study presents a design method to minimize the radiated sound power from vibrating beams and plates. The method relies on altering the acoustic characteristics of beam and plate structures passively by forming dimples and beads on their surfaces. The dimples and beads change the local stiffness of the structure without changing its mass. Also, they alter the mode shapes so as to reduce sound

power. The vibration response of the dimpled and beaded beams and plates are calculated using the finite element method (i.e., ANSYS parametric design language). Then, the radiated sound power from vibrating structure is calculated based on the Lumped Parameter Model (LPM). Finally, the method of Genetic Algorithm (GA) is used to optimally locate and size the dimples or the beads on the plate to minimize the sound radiation. The sound radiation is minimized either at a single frequency, or over a broad frequency band. The results show that dimples or beads forming on thin beams and plates can achieve effective reductions in radiated sound power.

10:55

3aSA3. Using optimization for acoustic cloak design. Liang-Wu Cai and Chunyan Bao (Mechanical and Nuclear Engineering, Kansas State University, Manhattan, KS 66506, cai@ksu.edu)

Many of the acoustic cloak designs are based on the acoustic analogy of transformation optics. This process dictates some major challenges when realizing such cloaks: the materials are very otherworldly. One of the most significant obstacles, the mass-anisotropy, has been removed by using two isotropic acoustic materials to create an equivalent anisotropy. In this presentation, optimization is used as a tool for designing acoustic cloaks such that materials can be feasibly fabricated using naturally occurring materials. The initial designs are based on Cummer-Schurig prescription for acoustic cloak, rendered in a layered structure, and using the anisotropic-isotropic equivalency. The first is using unconstrained optimization to demonstrate that material singularity is not a requirement for perfect cloaking. The optimization is able to fine-tune material properties in the initial design to achieve perfect cloaking within a limited frequency range. The second work is using multi-objective optimization to expand the range of the frequency range in which the cloaking remains effective. The third are is to use constrained optimization to limit the material properties to ranges that are naturally available. Lastly, different optimization techniques are combined to design acoustic cloaks that are made of mixtures of conventional isotropic solid and fluid (acoustic) materials.

Contributed Papers

11:15

3aSA4. Fast wave source localization with sparse measurements. Anthony Sabelli and Wilkins Aquino (Duke University, Durham, NC 27705, ajsabelli@gmail.com)

Source localization problems are encountered in a variety of engineering dis- ciplines. Applications include earthquake localization, damage identification, speaker localization and structural testing. In most realistic settings, measure- ment points are sparse with respect to the physical domain. Moreover, the experimenter may not have control over where to place measurement points. It is in these imperfect settings that we still need to estimate the location of a wave source. In this talk we will outline a method for source localization inspired by the topological derivative used in shape identification. We will draw parallels to phase conjugation mirror techniques and gradient optimization. We will make no assumptions about the nature of the ambient media nor the computational domain. Specifically we allow for energy loss. We will also outline implemen- tation within an existing massively parallel finite element solver. Our proposed algorithm is minimally invasive and fully exploits the underlying optimization in the existing solvers. Moreover we can extend the method to other physical contexts.

11:30

3aSA5. Inverse acoustic source identification in a massively parallel finite element framework. Timothy Walsh (Computational Solid Mechanics and Structural Dynamics, Sandia National Laboratories, PO Box 5800, MS 0380, Albuquerque, NM 87185, tfwalsh@sandia.gov), Wilkins Aquino (Civil and Environmental Engineering, Duke University, Durham, NC), Denis Ridzal, and Joe Young (Uncertainty Quantification and Optimization, Sandia National Laboratories, Albuquerque, NM)

Characterizing the frequency spectrums of acoustic sources from measured accelerometer or microphone data is a common inverse problem in engineering acoustics. Applications include acoustic testing of aerospace structures, room acoustics, and underwater acoustics. Typically, accelerometer or microphone pressures are measured, and it is desired to characterize the acoustic sources that produced these measurements. Many of these applications of interest involve large acoustic domains and high frequency ranges, thus making a finite element solution an attractive option for the forward problem. In this talk we will present a partial differential equation (PDE) constrained optimization approach for solving the inverse problem that is based on a coupling between a finite element-based massively parallel structural dynamics code (Sierra-SD) and a massively parallel optimization code (Rapid Optimization Library (ROL)). Gradients and solution iterates are exchanged between the codes during the solution process. The gradients for the optimization solver are computed using the adjoint method, which translates to forward and adjoint Helmholtz solves in the frequency domain. We will present results on several problems of interest. Sandia is a multiprogram engineering and science laboratory operated by Sandia Corporation, a Lockheed Martin Company, for the US Department of Energy's National Nuclear Security Administration. (DE-AC04-94AL85000)

11:45

3aSA6. Sound radiation of double rotating dipole. John Wang and Hongan Xu (Volvo Construction Equipment, 312 Volvo Way, Shippensburg, PA 17257, john.wang@volvo.com)

Mechanical devices are modeled as single and double rotating dipoles in this study. The expressions for sound pressure, intensity and power are derived. Sound pressure and sound power level reduction is investigated. The reduction is quantified. The theory is compared with test data. The applications are discussed.

Session 3aSC

Speech Communication: Speech Production I: Segments and Suprasegmentals (Poster Session)

Jie Zhang, Chair
Linguistics, The University of Kansas, Lawrence, KS 66045

Contributed Papers

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

3aSC1. Speech production under real-time simulation of cochlear implant acoustic feedback. Elizabeth D. Casserly (Dept. of Linguistics, Indiana University, Memorial Hall Rm 322, Bloomington, IN 47406, casserly@indiana.edu) and David B. Pisoni (Dept. of Psychological & Brain Sciences, Indiana University, Bloomington, IN)

Although previous research on simulation of cochlear implant (CI) processing with normal-hearing listeners relied on offline transformation of prerecorded acoustic signals, the advent of new simulation technology using a Portable Real-Time Vocoder (PRTV) enables subjects to experience CI simulation not only of interlocutors' speech during face-to-face interaction, but also of their own speech acoustics. This paper explores the effects of this novel acoustic feedback manipulation on subjects' speech production. Nine normal-hearing speakers were recorded producing 114 isolated English words and 24 sentences during three time epochs: once under normal conditions, once immediately after being fitted with the PRTV, and again after experiencing a short, natural conversational interaction through the real-time (8-channel, noise-vocoded) CI simulation. Acoustic-phonetic analysis of subjects' speech revealed substantial, segment-specific shifts in vowel acoustics, alteration of sibilant frication frequencies, and evidence of increased cognitive load (e.g. slower speaking rates) during speech production under conditions of vocoded acoustic feedback. Speakers also appeared to alter their articulatory strategies, spending more time on production of consonants relative to vowels, possibly reflecting a flexible exploitation of reliable somatosensory versus aural feedback cues.

3aSC2. Developmental changes in voiceless fricative production: Influence of position in words. Kanae Nishi and Elizabeth C. Graham (Boys Town National Research Hospital, 555 N. 30th Street, Omaha, NE 68131, kanae.nishi@boystown.org)

Children master the production of fricative consonants later than other phonemes [Moeller et al., Ear Hear. 28, 605-627 (2007)]. Even though recognizable fricative categories are present before school age, fine-tuning of acoustic properties may continue throughout early adolescence [Nissen & Fox, J. Acoust. Soc. Am. 118, 2570-2578 (2005)]. Previous acoustic studies on the development of fricative production focused on those in word-initial position only. Even in adults' speech, acoustics of consonants in word-medial or word-final positions vary more compared to those in the word-initial position. The present study hypothesized that adult-like production of fricatives in the word-final position may be achieved later than those in the word-initial position due to the acoustic variability in the adult exemplars children hear. Thirty-six (six each of 4-, 6-, 8-, 10-, 12-year-olds and adults) female native speakers of American English recorded five tokens of 16 consonant-vowel-consonant monosyllabic real words containing voiceless fricative consonants /f θ s ʃ/ in initial or final position in /i/ and /ɑ/ vowel contexts. Each token was analyzed for frication duration, amplitude, and several spectral measures. Results will be discussed in terms of fricative position in word, vowel context, and speaker age. [Work supported by NIDCD R03 DC009334 and P30 DC004662.]

3aSC3. Effects of articulatory planning factors on children's production of plural *-s*. Rachel M. Theodore (University of Connecticut, 850 Bolton Road, Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu), Katherine Demuth (Macquarie University, Sydney, NSW, Australia), and Stefanie Shattuck-Hufnagel (Massachusetts Institute of Technology, Cambridge, MA)

Children's early use of grammatical morphemes is notoriously variable. Recent findings indicate that some variability in early productions is systematically related to speech planning factors, suggesting that variability in morpheme production is not solely the consequence of impoverished syntactic representations. For example, research has shown that plural *-s* is produced more reliably in utterance-final compared to utterance-medial position. Here we examined the locus of the positional effect for plural *-s*. Productions of eight plural nouns in utterance-medial and utterance-final position were elicited from three groups of 2-year-olds. Across the groups, we manipulated articulatory difficulty of the medial context such that it consisted of a stop consonant (e.g., dogs bark), a stressed vowel (e.g., dogs eat), or an unstressed vowel (e.g., dogs arrive). Results showed a robust positional effect for the difficult context created by the stop consonant. The positional effect was not observed for the simple articulatory context created by the stressed vowel. However, planning difficulty for the unstressed vowel was sufficiently increased such that the positional effect again emerged in this context. These results suggest that production of grammatical morphemes is influenced by articulatory planning factors, which points to specific constraints for theoretical accounts of language acquisition.

3aSC4. Understanding speech acoustics in an era of extreme cue-integration: Multi-dimensional phonetics reveals individual differences in fricative production. Ariane E. Rhone (Neurosurgery, University of Iowa Hospitals and Clinics, E11 Seashore Hall, Dept of Psychology, Iowa City, IA 52242, ariane-rhone@uiowa.edu), Keith S. Apfelbaum, and Bob McMurray (Psychology, University of Iowa, Iowa City, IA)

Phonological features are indicated by many acoustic cues (Lisker, 1986). Listeners must thus combine multiple cues to recognize speech (Nearey, 1990). A recent survey of cues to English fricatives identified 24 distinct, useful cues (McMurray & Jongman, 2011). This multi-dimensional nature of speech raises methodological challenges: cues do not contribute equally, and multiple cues contribute to the same phonetic features (e.g. voicing). We offer a solution, using McMurray and Jongman's (2011) fricative database as a test. We used a logistic regression to predict the fricatives using measurements of 24 cues. We then sum the product of each cue value and its weight from the regression to determine the strength of the evidence for a given feature/phoneme (e.g. degree of "f-ness" vs. "v-ness") for each token. By computing this for different subsets of cues, we can measure how classes of cues work in tandem. We illustrate this by examining the relative contribution of cues within the frication and those within the vocoid, as well as spectral and temporal cues, to show individual differences in fricative productions. These analyses offer a straightforward approach to conceptualizing speech perception in high multi-dimensional space, while also giving insights on the production of English fricatives.

3aSC5. Linguistic effects on the timing of gestural coordination in Modern Greek consonant clusters. Jonathan C. Yip (Linguistics, University of Michigan, 455 Lorch Hall, 611 Tappan Street, Ann Arbor, MI 48109, jonyip@umich.edu)

Although position in word, order of place of articulation, and manner of articulation have been shown to influence gestural timing in CC clusters in various languages, there is no consensus on whether these timing patterns are due to biomechanical or perceptual-recoverability constraints (or both) on production. This study of Modern Greek speakers' CC productions investigates the effects of within-word position (initial, medial), place order (front-to-back, back-to-front), C1 manner (plosive, fricative), and C2 manner (plosive, fricative, liquid) on articulatory lag between C1 and C2. A perception-oriented account predicts an influence of both C1 and C2 manner on gestural lag when acoustic masking is most likely (i.e. when CC is word-initial and back-to-front), whereas a biomechanical account predicts no such effect. To assess relative degree of gestural lag, ultrasound imaging and lip video are used to track the timing patterns of labial, tongue-tip, and tongue-dorsum constrictions during production. Preliminary data on word-initial CC sequences show clear gestural onset lag and achievement lag in [kt] relative to [ks] and [kl], but no onset or achievement lag in [pt] relative to [ps] and [pl], consistent with the perceptual-recoverability hypothesis.

3aSC6. Prosodic position effects on the statistical relationships between distinctive features and acoustic-phonetic properties of English consonants. Noah H. Silbert (Center for Advanced Study of Language, University of Maryland, 7005 52nd Ave, College Park, MD 20742, nsilbert@umd.edu), Kenneth J. de Jong, Kirsten Regier, and Aaron Albin (Linguistics, Indiana University, Bloomington, IN)

Previous research has shown that distinctive features must interact extensively to account for the location and shape of phonological consonant categories in multidimensional acoustic space (de Jong, et al., 161st ASA Meeting). The current analysis focuses on how syllable position (onset vs. coda) modulates feature interactions in the consonants /p, b, t, d, f, v, s, z/. Statistical model comparisons indicate that models allowing pervasive interactions between features and syllable position fit better than do more restrictive models with few or no interactions. Some interactions between syllable position and features are well-documented, such as vowel duration distinguishing voicing more robustly in coda position than in onset position. Other such interactions are novel. For example, consonant duration can cue both voicing and manner contrasts, with duration differences corresponding more strongly to manner contrasts in onset position and more strongly to voicing contrasts in coda position. Similarly, measures of noise power distinguish coronals from labials in onset position, whereas place and voicing interact in codas. These results contribute to a picture of the acoustic distribution of consonants being not only segment-specific, but also determined substantially by the position of the consonant within a syllable.

3aSC7. Phonetic effects of distance in Burmese. Becky Butler (Cornell University, 203 Morrill Hall, 159 Central Ave., Ithaca, NY 14853, bbt24@cornell.edu)

This study investigates the phonetic effects of distance from phonological boundaries. Burmese, a Sino-Tibetan language, has a set of words of the shape Cə.Cə.(CV), in which the last syllable is footed, may contain any vowel in the inventory, and carries tone; whereas unfooted syllables contain only the vowel [ə] and do not have lexical tone (Green 1995, 2005). Under a purely phonological interpretation, we may expect all unfooted syllables to be identical in terms of duration and vowel quality. However, Chitoran and Hualde (2007) show that for Romance languages, distance from stress - a phonological entity - can cause significant durational differences between pretonic and pre-pretonic vowels. Similarly, the present study finds that although formant values between prefooted and pre-prefooted vowels in Burmese are not significantly distinct for any speakers - suggesting that vowel quality in all unfooted syllables is similar - distance from the tone-bearing syllable causes significant durational differences for three of five speakers ($p < 0.0001$). These results suggest that the data can be explained by neither phonological categorality nor phonetic gradience alone, but that both play a role in speech production and that the importance of each varies across speakers.

3aSC8. Language specificity in the perception of children's productions of /t/ and /k/. Benjamin Munson (Speech-Language-Hearing Sciences, University of Minnesota, 115 Shevlin Hall, 164 Pillsbury Drive SE, Minneapolis, MN 55455, munso005@umn.edu), Kiyoko Yoneyama (Department of English, Daito Bunka University, Tokyo, Kanto, Japan), and Jan Edwards (Communicative Disorders, University of Wisconsin, Madison, WI)

The age and order of acquisition of what are ostensibly the 'same' sounds can differ across languages. These differences relate to a number of factors, including frequency in the ambient language, language-specific phonetic instantiations of sounds, and language-specific parsing of children's emerging productions (Beckman & Edwards, 2010; Edwards & Beckman, 2008; Li et al, 2011). The current investigation examines the role of adults' perception of children's speech on the acquisition of /t/ and /k/ in English- and Japanese-speaking children. Previous work has shown that /t/ is acquired earlier than /k/ in English, but that the opposite is true in Japanese (Nakanishi et al., 1972; Smit et al., 1990). We examined whether this tendency was due to cross-linguistic differences in adults' perception of English- and Japanese-acquiring children's speech. Native speakers of English and Japanese labeled a large set of 2- to 5-year-old children's word-initial /t/- and /k/ productions. Japanese-speaking adults perceived English-speaking children's productions of sounds intermediate between /t/ and /k/ as more /k/-like than did English-speaking adults. This suggests that the earlier acquisition of /k/ in Japanese than in English may be due, in part, to Japanese-speaking adults' willingness to label ambiguous sounds as /k/-like.

3aSC9. Consonant-f_0 interaction under predictable voicing: f_0 lowering due to phonetic factors. Indranil Dutta (Computational Linguistics, English and Foreign Languages University, Department of Computational Linguistics, Tarnaka, Osmania University Campus, Hyderabad 500605, India, indranil.dutta.id@gmail.com), Jasmine M. George, and Minu S. Paul (Language Sciences, English and Foreign Languages University, Hyderabad, Andhra Pradesh, India)

We report on consonant-f_0 interactions in Malayalam. Crosslinguistically, voicing lowers f_0 in the following vowel (House and Fairbanks 1953, Hombert 1978, Clements 2002, Moreton 2006). While this lowering has been attributed to physiological and phonetic factors (Stevens 1998, Atkinson 1978, and Honda 2004), Ohde (1984), Svantesson and House (2006), and Keating (1984) have argued that f_0 lowering serves to maintain a phonological contrast between voiced and voiceless segments. Voicing in Malayalam is predictable; voiceless stops appear initially, and voiced intervocalically. We report on data from 6 native speakers. 3 repetitions of each word in a frame sentence were recorded. Since Malayalam words only appear with voiceless initial stops, we contrast these with nasals from corresponding places of articulation. We examine time-normalized f_0 perturbation due to voicing in Malayalam for all places of articulation for both initial and medial contexts. Our findings lend support to the physiological and phonetic account of f_0 lowering. Malayalam exhibits a pattern of f_0 lowering following voicing and raising during the production of voiceless segments. In spite of Malayalam voicing being predictable, the f_0 perturbation in the following vowel follows the cross-linguistic pattern of lowering following voiced segments. This finding dove-tails with the results from earlier studies that take physiological factors to be responsible for the lowering of f_0 following voiced segments.

3aSC10. Acoustic correlates of breathy voice in Marathi sonorants. Kelly Berkson (Linguistics, University of Kansas, Lawrence, KS 66049, keberkson@gmail.com)

Breathy voiced sonorants occur in fewer than 1% of the languages indexed in the UCLA Phonological Segment Inventory Database. Acoustic analysis of these sounds remains sparse, and our understanding of the acoustic correlates of breathy voice in sonorants is incomplete. The current study presents data from Marathi, an Indo-Aryan language which boasts a number of breathy voiced sonorants. Ten native speakers (five male, five female) were recorded producing Marathi words embedded in a carrier sentence. Tokens included plain and breathy voiced nasals, laterals, rhotics, and approximants before the vowel [a]. Measures reported for consonants and subsequent vowels include duration, F0, Cepstral Peak Prominence (CPP), and corrected H1-H2*, H1-A1*, H1-A2*, and H1-A3* values. As expected, breathy sounds have lower CPP values than modal sounds, and larger positive values for the remaining spectral measures. The spectral effect of breathiness extends from the beginning of the consonant through the end of the vowel. While some breathy voiced sounds contain a salient breathy interval that is highly visible in the waveform and spectrogram, others don't, and in its absence the spectral differences between breathy and modal sounds are greatly increased.

3aSC11. Acoustic and aerodynamic characteristics of nasal diphthongs in Brazilian Portuguese. Rita Demasi (Linguística, USP, Rua Conde de Itu, 804, São Paulo, São Paulo 04741-001, Brazil, ritademasi@gmail.com) and Didier Demolin (GIPSA-LAB, Université Stendhal, Grenoble, Rhones-Alpes/Grenoble, France)

Previous studies describe acoustic and aerodynamic aspects of nasalized vowels in Brazilian Portuguese. However, there are few studies characterizing nasal diphthongs in this language. The aim of this work is to analyze acoustic and aerodynamic characteristics of nasal diphthongs of Brazilian Portuguese spoken in the city of São Paulo. We compared oral and nasal diphthongs to identify the main features of these sounds and to understand the timing of velum movements. Our data was recorded with the Portable EVA 2 workstation. The corpus of this experiment was made of ten oral and ten nasal diphthongs, with back and front glides at the end: /aw/ and /ej/; /ãw̃/ and /ẽj̃/. Words were inserted in the following sentence [ʤi.gʊ___kadɐ ʤiɐ] and [ʤi.gʊ___todʊ ʤiɐ]. The first phrase was repeated three times with six subjects and the second with three subjects. The corpus was analyzed with the Signal Explorer and Phonédit Softwares. The aerodynamic parameters analyzed were duration, peak and volume of nasal airflow. The acoustic parameters analyzed were formants patterns and FFT spectrum. Aerodynamic and acoustic data show that in nasalized diphthongs, the nasalized vowel is followed by a nasal glide and a short nasal appendix. The peak of nasal airflow at the end of the nasal diphthong is due to the fact that the oral closure, which is made for the following voiceless stop, is produced with an open velum.

3aSC12. Voicing, aspiration, and vowel duration in Hindi. Karthik Durvasula and Qian Luo (Michigan State University, East Lansing, MI 48824, durvasul@msu.edu)

There is extensive evidence that consonantal laryngeal features modulate adjacent vowel duration (Chen 1970). However, it is not clear if both consonant voicing and aspiration affect vowel duration. Previous studies (on Hindi) produced inconsistent results with respect to the effect of consonant aspiration on vowel duration, while finding a clear positive correlation with consonant voicing (Maddieson & Gandour 1976; Ohala & Ohala 1992; Lampp & Reklis 2004). We conducted an experiment on 7 native standard Hindi speakers, who produced 10 repetitions of 12 nonsense words ending in [d̪, d̪ʰ, t̪, t̪ʰ] that had 3 different CVCVV contexts. The results of the experiment show that there is a statistically significant main effect of coda voicing on vowel duration and a marginally significant main effect of aspiration on vowel duration. Furthermore, the effect of the aspirated coda consonants on vowel duration appears to be modulated by the surrounding segmental context. The results suggest that both consonant voicing and aspiration increase adjacent vowel duration. The results also suggest that the inconsistent findings in previous studies with respect to the effect of aspiration could be the result of differing phonetic contexts of the relevant segments.

3aSC13. Hierarchical Bayesian modeling of vowel formant data: Speaker-intrinsic and speaker-extrinsic approaches compared. Aaron L. Albin and Wil A. Rankinen (Department of Linguistics, Indiana University, Memorial Hall, 1021 E 3rd St, Bloomington, IN 47405-7005, aaalbin@indiana.edu)

Vowel formant data is traditionally normalized across speakers by transforming a set of 'raw' measurements into 'standardized' ones in one of two ways. With a *speaker-extrinsic* method, data from each individual is normalized with respect to external baseline measures calculated across the population of all speakers in a corpus, whereas a *speaker-intrinsic* method normalizes entirely with respect to speaker-dependent variables. The present study reports on implementations of both these methods in terms of hierarchical statistical models whereby probability distributions for various model parameters can be obtained using Bayesian analysis (rather than merely 'converting' the measurements). In this new framework, a speaker-extrinsic approach can estimate (1) the size and shape of each speaker's vowel space, (2) the locations of vowel categories across a speech community within a normalized space, and (3) individual speakers' deviations from the community norms. However, this process relies on a number of assumptions that are not needed with a speaker-intrinsic approach, which instead makes many low-level discrete 'decisions' on a speaker-by-speaker basis. By testing multiple models on the same dataset (a large corpus of vowel data collected from 132 speakers of American English), the present study explores the comparative merits of speaker-extrinsic and speaker-intrinsic Bayesian models.

3aSC14. The effect of physical appearance and accent on the elicitation of vowel hyperarticulation by British English native speakers in speech to foreigners. Jayanthiny Kangatharan (School of Social Sciences, Brunel University, 36 Abbots Drive, London, Harrow HA2 0RE, United Kingdom, hspgjjk@brunel.ac.uk), Maria Uther (School of Psychology, Univ. of New South Wales, London, Uxbridge, United Kingdom), and Fernand Gobet (School of Social Sciences, Brunel University, London, Uxbridge, United Kingdom)

Speech aimed at infants and foreigners has been reported to include the physical exaggeration of vowels, that is vowel hyperarticulation. Although infants have been demonstrated to experience hyperarticulated vowels in speech directed at them, little research has been done on whether vowel hyperarticulation occurs as a result of foreign appearance, foreign accent or as a consequence of both looking and sounding foreign. The present study explored if appearance and speech separately affect the native speakers' hyperarticulation. Fifty-two White British adult speakers communicated with one of four different confederate groups (2 types of appearance x 2 types of accent) to solve three modified versions of the DiapixUK tasks. Results indicate that not appearance but speech had an effect on native speakers' production of vowels. Specifically, vowel space was significantly larger in speech directed to foreign-accented individuals than to individuals with native accent irrespective of their physical appearance. The acquired samples of hyperarticulatory speech will be used in perceptual identification and clarity tasks to ascertain which speech samples help native speakers to understand speech better.

3aSC15. Dialectal and age-related acoustic variation in vowels in spontaneous speech. Ewa Jacewicz and Robert A. Fox (Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, jacewicz.1@osu.edu)

Our knowledge of the acoustic characteristics of vowels is based mainly on productions obtained under fine experimental control (e.g., [hVd] tokens produced in isolation). Analyzing spontaneous speech data has its obvious challenges because the vowels of interest occur in various segmental and prosodic contexts and are additionally affected by sudden changes in speech tempo. These variations may compromise the accuracy of interpretation of linguistic phenomena such as sound change. In this study we examine if more natural productions of vowels in two dialects (as compared to those produced in citation form and read speech) show evidence of the existence of cross-generational vowel changes—including corresponding changes in spectral dynamics. A subset of vowels from a large corpus of spontaneous conversations was analyzed. The vowels occurred in variable consonantal contexts in both mono- and polysyllabic words. The obtained patterns of vowel change were consistent with those in read and citation form speech. Although the measured spectral changes were smaller due to shorter vowel durations, the dialect-specific nature of formant dynamics was maintained. These results demonstrate that the patterns of vowel change, including variation in formant dynamics, do not diminish under the circumstances of greater contextual/prosodic variability found in spontaneous speech.

3aSC16. Cross-dialectal and cross-generational changes in point-vowel locations in English. Robert A. Fox and Ewa Jacewicz (Speech and Hearing Science, The Ohio State University, 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

The positions of individual vowels in the acoustic vowel space often change over time in languages such as English. This study examines the changes in the location of four point vowels [i u ɑ æ] for three English dialects (North, Midland and Inland South) as produced by three generations of speakers (children,

adults aged 35-50 and 65+ years). We determined the speaker-specific spectral centroids of the midpoints of a set of 12 monophthongs in the F1 x F2 vowel space produced by each individual speaker. These formant values were then normalized using Nearey's formula. For both the raw and normalized F1 x F2 spaces, polar coordinates for productions of each point vowel were calculated with this centroid as the origin (producing a radius and an angle—a variation of Chung et al.'s [2012]) along with vowel space areas. Compass plots were drawn for the vowels of each speaker and angle (polar) histograms for each vowel were created for each speaker group. Analysis of the histograms and radii showed significant differences among the groups in terms of the location of /æ/ and /u/ as well as area differences. The implications these data have for sound change in progress will be discussed.

3aSC17. Relationship between articulatory acoustic vowel space and articulatory kinematic vowel space. Jimin Lee and Susan Shaiman (Department of Communication Science and Disorders, University of Pittsburgh, Pittsburgh, PA 15206, jiminlee@pitt.edu)

The current study examines the relationship between articulatory acoustic vowel space and kinematic vowel space with an emphasis on the range of tongue movement by utilizing electromagnetic articulography. Subject population is 20 healthy female speakers. Electromagnetic articulography (AG-200) and a synchronized separate digital audio recording system were utilized to obtain kinematic and high quality acoustic data. Three coils on the tongue (tip, body, and dorsum) and one coil on the lower lip were used. To examine both intra- and inter- speaker relationship between articulatory acoustic and kinematic spaces, speech samples of ten different tasks that elicited various articulatory space sizes were collected. Each speaker produced three repetitions of four corner vowels in /h/-vowel-/d/ and /d/-vowel-/d/ consonant environments embedded in a carrier phrase in five different speaking styles (habitual, fast, slow, loud, and soft). Articulatory working space was generated from coordinates of first and second formant frequencies from acoustic data, and position X and Y from each coil kinematic data. Both acoustic and kinematic coordinate data are obtained at the same time sampling point. Results will be discussed in terms of amount of variance of acoustic vowel space explained by kinematic articulatory space and issues of interpretation of acoustic vowel space.

3aSC18. Predictability effects on vowel realization in spontaneous speech. Michael McAuliffe and Molly Babel (Linguistics, University of British Columbia, Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mcauliff@interchange.ubc.ca)

Previous research on vowel realizations within the formant space has found effects for lexical factors such as word frequency in both laboratory settings (Wright, 2004; Munson & Solomon, 2004; and others) and in spontaneous speech (Gahl, Yao, & Johnson, 2012). In addition to lexical factors, semantic context has also been found to influence vowel realizations in laboratory settings, such as emphatic/non-emphatic contexts (Fox & Jacewicz, 2009) and whether a word is predictable from the preceding words (Clopper & Pierrehumbert, 2008). The current project looks at whether effects on vowel realization for semantic context from the laboratory can be extended to spontaneous speech. As in Gahl, Yao, and Johnson (2012), the Buckeye Corpus (Pitt et al., 2007) will be used, with the same predictors used there with the addition of a semantic predictability measure. Semantic predictability for a given word will be calculated based on relatedness of that word to words five seconds before the word or less, where relatedness will be calculated based on WordNet (Princeton University, accessed 2012). As a listener can rely more on context for disambiguation, words that are predictable from their preceding context are hypothesized to contain less distinct vowels than words that are not predictable from context.

3aSC19. Acoustic correlates of vowel intelligibility in clear and conversational speech for young normal-hearing and elderly hearing-impaired listeners. Sarah H. Ferguson (Communication Sciences and Disorders, University of Utah, 390 South 1530 East, Room 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu) and Hugo Quene (Utrecht Institute of Linguistics OTS, Utrecht University, Utrecht, Netherlands)

Previous reports on the relationship between clear speech acoustic changes and the clear speech intelligibility benefit for vowels have used an "extreme groups" design, comparing talkers who produced a large clear speech benefit to talkers who produced little or no clear speech benefit. In Ferguson and Kewley-Port (2007), 12 talkers from the Ferguson Clear Speech Database (Ferguson, 2004) were assigned to groups based on the vowel identification performance of young normal-hearing listeners, while Ferguson (2010) chose 20 talkers based on the performance of elderly hearing-impaired listeners. The present investigation is employing mixed-effects models to examine relationships among acoustic and perceptual data obtained for vowels produced by all 41 talkers of the Ferguson database. Acoustic data for the 1640 vowel tokens (41 talkers X 10 vowels X 2 tokens X two speaking styles) include vowel duration, vowel space, and several different measures of dynamic formant movement. Perceptual data consist of vowel intelligibility in noise as reflected by the performance of young normal-hearing and elderly hearing-impaired listeners. Analyses will explore the relative importance of the various clear speech acoustic changes to the clear speech vowel intelligibility effect as well as the degree to which this relationship varies between the two listener groups.

3aSC20. Acoustic and physiologic measures of register transitions sung by females. Richard J. Morris (Communication Science and Disorders, Florida State University, 201 West Bloxham Road, 612 Warren Building, Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu), David Okerlund (College of Music, Florida State University, Tallahassee, FL), and Claire E. Dolly (Communication Science and Disorders, Florida State University, Tallahassee, FL)

The purpose of this study was to examine the physiological and acoustical adjustments made by trained female singers to transition across vocal registers when singing a chord triad. Ten female singers participated in this study. A microphone was placed 30 cm from the corner of the subjects mouth and EGG electrodes were positioned over their thyroid laminae. Audio and EGG signals were channeled to Voce Vista Pro software. Each singer was presented the starting pitch of A3 or 220 Hz, by the experimenter and directed to sing a chord triad using an [a:] vowel. Each triad was sung in a single breath at an andante tempo. The EGG measurements included the closing quotient (CQ) and the FFT spectrum measurements include the harmonic number, frequency, and amplitude of the harmonic with the greatest amplitude. Four singers lowered CQ, two raised CQ, and four maintained a steady CQ when changing vocal register. The frequencies and amplitudes of the peak harmonic remained fairly steady across the register shift. These changes indicated that the singers were tracking their vocal tract resonance to their CQ. The more trained singers used the third harmonic in their chest register and the second harmonic in middle register.

3aSC21. An acoustic analysis of lexical stress in Uyghur. Mahire Yakup (Linguistics, University of Kansas, Lawrence, KS 66044, mylyakup@ku.edu)

In this study, the accent pattern in Uyghur, a Turkic language, was investigated. Two experiments provide a detailed phonetic analysis in order to determine the acoustic cues to stress in Uyghur. In Experiment 1, six disyllabic minimal pairs (e.g., A-cha, a-CHA), contrasting in location of stress, were produced by five native Uyghur speakers with three repetitions in a fixed sentence context. In order to generalize the results from the small set of minimal pairs in the first experiment, Experiment 2 examined the initial syllable of disyllabic nouns that contrasted in first-syllable stress (e.g., DA-ka, da-LA) while syllabic structure (CV versus CVC) was also manipulated. In both experiments, average fundamental frequency, duration, and average intensity were collected in the vowels in accented and unaccented syllables. The results from both experiments showed that there were significant differences in duration and intensity between stressed and unstressed syllables, with the intensity differences moderated by syllable structure. No differences were found in fundamental frequency. While previous studies have classified Uyghur as a pitch-accent and a stress-accent language, the present acoustic data suggest that native speakers make no use of pitch cues to signal stress in Uyghur.

3aSC22. The effect of segmental makeup on Mandarin tone coarticulation. Yuwen Lai and Hui-Ting Huang (Foreign Languages and Literatures, National Chiao-Tung University, Taiwan, 1001 University Road, Hsinchu 30010, Taiwan, yuwen.lai@gmail.com)

The effect of segmental make-ups on tonal coarticulation was investigated in Mandarin. Target syllables differ in coda (no coda, alveolar nasal or velar nasal) and initial sonorancy (obstruent or sonorant) were embedded in trisyllabic non-words templates, in the form of trisyllabic names. Twenty native speakers were recorded and the F0 contours of all syllables were measured at a 10-ms time step. The coarticulation effect of four Mandarin tones are examined in three different positions with 4 (initial position), 16 (medial position), and 4 (final position) tonal contexts. Moreover, the phonetic variations of each tone in different prosodic positions were examined. The preliminary results indicate that the coarticulation effect is less prominent when intervened by obstruents whereas nasal codas amplify the effect. The realization of this modulating effect on the directionality of tonal coarticulation (carryover and anticipatory) will be discussed. The magnitude of coarticulation in compatible (adjacent tones have similar registers) and conflicting (adjacent tones with large register difference) environments will also be compared.

3aSC23. Acoustic differences in adult-directed and child-directed monosyllabic Mandarin tone productions. Puisan Wong, Xin Yu, Guanjin Zhang, Jiulin Zhu, and Tina Yeung (Otolaryngology–Head and Neck Surgery, The Ohio State University, 915 Olentangy River Road, Columbus, OH 43212, pswResearch@gmail.com)

To investigate the acoustic differences in adult- and child-directed monosyllabic Mandarin tone productions, twenty-one mothers of preschool children labeled pictures that represented monosyllabic words to their children and to another adult. Their productions were low-pass filtered to eliminate lexical information. Five judges determined the target tones based on the filtered stimuli. Acoustic analyses were performed on the productions in which the target tones were correctly identified by four or more of the judges. Preliminary results showed no duration difference in the four tones in adult-directed and child-directed productions. Overall, all the four tones were produced at significantly higher f0s in child-directed productions than in adult-directed productions. Specifically, child-directed Tone 1 productions were produced at significantly higher f0s, and the f0 contours exhibited higher positive f0 slopes and were not as level as in adult-directed Tone 1 productions. Child-directed Tone 2 productions were produced at higher f0s, spanned at larger f0 ranges but maintained the same rising slopes as in adult-directed productions. Child-directed Tone 3 and Tone 4 productions had the same f0 shapes as in adult-directed productions but were produced at higher f0s (Work supported by NIDCD).

3aSC24. A production and perception study on tonal neutralization in Nanchang Chinese. Jiang Liu and Jie Zhang (Linguistics, The University of Kansas, 1541 Lilac Lane, Lawrence, KS 66044, liujiang@ku.edu)

In a production study of tonal contrasts in lexically stressed but grammatically stressless syllables vs. lexically stressless syllables in Nanchang, a Gan dialect spoken in southeastern China, we found that tonal neutralization only occurs in lexically stressless syllables. We argue that the main phonetic ground for such a tonal contrast distribution lies in the rhyme duration difference between syllables with and without lexical stress, namely, lexically stressless syllables have shorter rhyme duration than lexically stressed but grammatically stressless syllables, and the shorter the rhyme duration of a syllable is the fewer tonal contrasts the syllable allows. In terms of perception, we found that different tonal contrasts indeed become neutralized in lexically stressless syllables. However, the neutralization pattern at the perception level is not the same as the one at the production level due to word specific effects.

3aSC25. Perception and production of Mandarin tones by English vocalists and instrumentalists. Shuang Lu, Joe Kirkham, Ratree Wayland, and Edith Kaan (Department of Linguistics, University of Florida, P.O. Box 115454, Gainesville, FL 32611-5454, shuanglu@ufl.edu)

Musical training has been found to positively affect non-native lexical tone perception and production (e.g. Wong et al., 2007). While there are comprehensive studies on trained instrumental musicians, relatively little is known about formally trained *vocal* musicians. The present study compares English vocalists and instrumentalists to see which type of musical training is more advantageous to lexical tone perception and production. Stimuli consisted of six syllables ([t^hi], [li], [mi], [t^ho], [lo], [mo]) associated with four Mandarin tones (high-level, high-rising, low dipping, and high falling). Native Mandarin non-musicians, native English non-musicians, vocalists, and instrumentalists (n = 15 per group) were tested in a same/different discrimination task and an imitation task. In the discrimination task, the English vocalists [d' = 3.12] performed significantly better than the English non-musicians [d' = 2.41; p = 0.04]. The vocalists also numerically outperformed the instrumentalists [d' = 2.84], who in turn outperformed the English non-musicians. Analyses on the "different" and "same" tone-pairs showed that the vocalists are marginally more accurate than the instrumentalists for T2-T2 pair-type only [p = 0.067]. In the imitation task, the three English groups did not differ in how their productions were evaluated by the native Mandarin judges.

3aSC26. Pitch and intensity in the speech of Japanese speakers' of English: Comparison with L1 speakers. Jun Okada, Ian L. Wilson, and Miyuki Yoshizawa (CLR Phonetics Lab, University of Aizu, Tsuruga, Ikki-machi, Aizuwakamatsu, Fukushima 965-8580, Japan, s1170182@u-aizu.ac.jp)

The speech of L2 learners of English is often difficult to understand because of intonation problems and misplaced word stress. In this research, we investigated whether or not the intonation patterns of Japanese speakers of English show common patterns based on proficiency level. First, we recorded "The North Wind and the Sun" from 50 Japanese undergraduate students (aged 18 to 24). We recorded native English speakers and also obtained such native data online. Next, we labeled each word and analyzed the pitch and intensity using Praat. Data was separated by gender and by proficiency in English, results were plotted, and statistical analysis was undertaken. Preliminary results show that pitch (and to a lesser extent, intensity) showed a common pattern across native speakers, but that L2 speakers relied on intensity much more than pitch in the production of stress.

3aSC27. Prosodic characteristics of three sentence types in Thai. Alif Silpachai (Linguistics, University of Southern California, Los Angeles, CA 90089, asilpachai@ucla.edu)

This study presents an acoustic analysis of three sentence types in Thai (declarative, interrogative, and emphatic) with the goal of providing a basic characterization of their prosody. To investigate prosodic realizations of sentence final syllables, we placed, in a sentence-final position, a target word which varied in one of the 5 lexical tones in Thai. We also varied the tonal context before the target word so that the pre-target word ends with low (21), mid (31), or high (45) tones. Preliminary results from one speaker show that F0 measures, especially f0 maximum, minimum, and range, differed across sentence types. In particular, emphatic sentences were distinguished from non-emphatic sentences by expanded F0 range, whereas target words in questions were distinguished from those in declarative sentences by both higher F0 maximum and minimum. Syllable duration also played a role in signaling emphasis and question: emphatic sentences were significantly longer than non-emphatic sentences, and questions were significantly shorter than declarative sentences. Interestingly, the tonal pattern of the target word changed for the case of emphasis when the target word had 31 and 45 tones. We will present findings from four additional Thai speakers and discuss their relevance to the intonational phonology of Thai.

3aSC28. Brazilian Portuguese intonation: A comparison between automatic and perceptual analyses. Waldemar Ferreira Netto, Marcus Vinicius Moreira Martins, Daniel Oliveira Peres, Renata Cezar de Moraes Rosa, and Joice Medeiros (Dept. de Letras Clássicas e Vernáculas, Universidade de São Paulo, Av. Professor Luciano Gualberto, 403, São Paulo 05508-900, Brazil, marcusvmmartins@gmail.com)

The present work aims to determine the most appropriated automatic analysis of phrasal intonation patterns in Brazilian Portuguese, understood as the variation of F0. In order to evaluate the automatic analysis done by the software ExProsodia a perceptual analysis was carried out and its results were compared to the ones given by the software. The corpus consisted of one sentence produced three times by 11 female subjects. In the perceptual analysis an intuitive evaluation was done considering the syllabic nuclei as intonation units (control group). The data from the automatic analysis were analyzed in three different ways: i) the cumulative mean of all points of intonation curve; ii) the cumulative mean of points higher than 0Hz; iii) and the cumulative mean of the points selected by ExProsodia. The acoustic parameters considered for the automatic analysis were: F0- from 50Hz to 700Hz; duration- from 20ms to 600ms; intensity- higher than 10% of RMS mean of intonation curve. A Dunnett's test compared the control group with other groups from

the automatic analysis ($\alpha = 95\%$, $n = 33$). In the comparison between the control group and i & ii, it was obtained $p < 0.05$. It was verified that the units established by the interval given by ExProsodia (iii) were the only ones that showed no significant differences when compared to the control group. The results indicate a similarity between the automatic and perceptual analyses. The automatic analysis done by ExProsodia seems thus trustworthy.

3aSC29. Language grouping based on pitch interval variability. Diana Stojanovic (Linguistics, University of Hawai'i at Manoa, 1811 East-West Rd., 933, Honolulu, HI 96848, dswongkaren@gmail.com)

Language rhythm has been discussed so far as a consequence of regularity, phonological differences, and in most recent literature as a result of durational variability. Original support coming from perception experiments is recently often questioned at least in terms of how well listeners can classify language samples stripped of all but durational information. In particular, pitch has been suggested to play a significant role in perceiving rhythmic differences. In Patel 2006, language and music rhythm differences between English and French were measured by means of durational and melodic measures. It was shown that the variability of pitches did not distinguish between two languages, but the variability of pitch excursions was significant. In this paper, we use pitch excursion variability as a measure applied to read samples of several languages. Preliminary results on 30 sec samples show that Brazilian Portuguese, French, Hawaiian, and Indonesian group together compared to English and German. The first group has more frequent but less prominent excursions (60 Hz) while the second group has less frequent but larger excursions (100Hz). Based on these results, originally proposed regularity can at least partly be explained by the variability of pitch intervals, grouping languages into "smooth" vs. "more prominent".

3aSC30. Discriminating languages with general measures of temporal regularity and spectral variance. Kathy M. Carbonell (Speech, Language & Hearing Sciences, University of Arizona, 1131 E 2nd St, Tucson, AZ 85721, kathyc@email.arizona.edu), Dan Brenner (Linguistics, University of Arizona, Tucson, AZ), and Andrew J. Lotto (Speech, Language & Hearing Sciences, University of Arizona, Tucson, AZ)

There has been a lot of recent interest in distinguishing languages based on their rhythmic differences. A common successful approach involves measures of relative durations and duration variability of vowels and consonants in utterances. Recent studies have shown that more general measures of temporal regularities in the amplitude envelope in separate frequency bands (the Envelope Modulation Spectrum) can reliably discriminate between English and Spanish [Carbonell et al. J. Acoust. Soc. Am. 129, 2680.]. In the current study, these temporal structure measures were supplemented with measures of the mean and variance of spectral energy in octave bands as well as with traditional linguistic measures. Using stepwise discriminant analysis and a set of productions from Japanese, Korean and Mandarin speakers, this suite of both acoustic and linguistic measures were tested together and pitted against each other to determine the most efficient discriminators of language. The results provide insight into what the traditional linguistic measures of speech rhythms are telling us about how language type structures the acoustic signal.

3aSC31. An experimental study of Korean rhythm structure on the basis of rhythm metrics. Eun-Sun Tark (Linguistics, University of Kansas, Lawrence, KS 66045, estark713@gmail.com)

This paper investigates the rhythm structure of Korean. Metrics used in this study included %V, ΔC, Varco V, nPVI-V, and rPVI-C, which have been shown to reflect differences in rhythm structure (stress-, syllable-, and mora-timing) across languages. Ten female native Koreans each produced 20 short declarative Korean sentences. The rhythm metric results of Korean were compared to those of English, Spanish, and Japanese using raw data from previous studies [Ramus et al. 1999, Cognition 73, 265-292; White and Mattys, 2007, Journal of Phonetics 35, 501-522; Grenon and White, 2008, BUCLD 32, 155-166]. Results indicate that Korean combines aspects of both syllable timing and mora timing. Korean has a similar ΔC, Varco V, and nPVI-V to syllable-timed languages. Korean has a similar %V and nPVI-V to mora-timed languages. These data show that instead of belonging to one of three distinct rhythm categories, languages may be placed along a rhythm structure continuum. On such a continuum, Korean is placed between syllable-timed and mora-timed languages, and distinct from stress-timed languages.

3aSC32. Phonetic characteristics of syllable reduction and enhancement in American English. Keiichi Tajima (Department of Psychology, Hosei University, Tokyo, Japan, tajima@hosei.ac.jp) and Stefanie Shattuck-Hufnagel (Speech Communication Group, Research Laboratory of Electronics, Massachusetts Institute of Technology, Cambridge, MA)

Syllables have been argued to play an important role in the prosodic organization of spoken language, but the number of syllables that speakers produce or listeners hear in an utterance is not always clear-cut, and may vary in subtle ways. For example, speakers may reduce or delete unstressed vowels in casual speech, e.g., producing *support* as *s'port*, on one hand, or insert vowels in careful speech, e.g., producing *please* as *puh-lease*, on the other. Relatedly, duration differences in a non-vowel segment may lead to changes in word identity based on the presence/absence of an unstressed vowel, as when different durations of /l/ in *blow* are sufficient to lead listeners to perceive either *blow* or *below*. The present study investigates how often such apparent changes in syllable count occur in spoken English, which phonetic contexts they tend to occur in, and to what extent these changes are probabilistic, by analyzing productions from a phonetically annotated corpus of conversational American English. Preliminary results suggest that unstressed vowels are often reduced enough to be omitted from the transcription, and this occurs more in certain phonetic contexts than others. Further results, and implications for theories of speech production and perception, will be discussed. [Work supported by JSPS.]

3aSC33. Video recordings of L1 and L2 jaw movement: Effect of syllable onset on jaw opening during syllable nucleus. Yusuke Abe, Ian L. Wilson (CLR Phonetics Lab, University of Aizu, Tsuruga, Ikkimachi, Aizuwakamatsu 965-8580, Japan, s1170175@gmail.com), and Donna Erickson (Showa Music University, Kawasaki, Kanagawa, Japan)

Video is a convenient, inexpensive method of recording data for jaw movement during speech. However, when using markers attached to the chin, it is possible that the data will not represent actual mandible motion, because of the skin stretching over the mandible - especially true for labial consonants. In this study, we made video recordings of L1 and L2 speakers of English saying 5 trials of 34 sentences each, and we automatically measured the distance between paper markers attached to the chin and glasses. We compared jaw opening during syllable nucleus for syllables with and without labial onsets, for L1 and L2 English speakers of various proficiencies. Although speakers must stretch the lower lip upwards for a labial constriction, preliminary results show that there are no statistically significant differences for any speaker's jaw opening during the nucleus of non-labial- versus labial-onset syllables. There is also very little intra-subject variation in the metrical structure (as measured by jaw opening) for a given sentence across trials. However, across-trial variability in the time between jaw movement peaks is a lot less for L1 than for L2, presumably because these L2 speakers have not yet mastered the metrical structure of English.

3aSC34. Acoustic contrastivity in soft, conversational, and loud speech. Yunjung Kim and Ali Beslin (Dept. of Comm Sci & Disor, Louisiana State University, Baton Rouge, LA 70803, ykim6@lsu.edu)

Increasing vocal effort is frequently used in clinical practice as a strategy to enhance speech intelligibility of individuals with various speech problems, particularly dysarthria. However, it is not straightforward why this strategy might yield better speech intelligibility, although some potential contributors have been suggested including hyperarticulation of vowels, greater presentation level of sounds (for consonant identification accuracy) or both. This presentation focuses on the change of *relative* contrastivity within utterances that were produced with gradually increasing vocal efforts (soft, conversational and loud, see Kim, ASA, 2011) to examine whether acoustic contrastivity is exaggerated with increasing vocal intensity and to identify the acoustic variables that produce contrastivity that are more or less sensitive to changes in vocal intensity. In this presentation, data on the ratio of vowel durations (long vs short), formant structures of vowels (tense vs lax) as well as the ratio of M1 of /s/ vs /ʃ/ will be compared across soft, conversational and loud speech conditions produced by young female adult speakers.

Session 3aUW

Underwater Acoustics and Signal Processing in Acoustics: Random Matrix Theory

Kathleen E. Wage, Cochair
George Mason University, 4400 University Dr., Fairfax, VA 22030

James C. Preisig, Cochair
WHOI, Woods Hole, MA 02540

Chair's Introduction—8:10

Invited Papers

8:15

3aUW1. Random matrix theory in signal processing: Performance analyses and algorithm design. Christ D. Richmond (MIT Lincoln Laboratory, 244 Wood Street, Lexington, MA 02420, christ@ll.mit.edu)

Estimation of covariance matrices from a finite sample set of data observations plays a central role in signal processing. The true data covariance is rarely known in practice, but optimized algorithms inevitably depend on such statistics to enable robust system performance; for example, environments plagued by dominant interference, and/or those challenged by complex propagation. Classical (finite) random matrix theory (RMT) facilitates assessment of finite sample effects surrounding covariance estimation. Implements shown to be very useful in this regard under a circular complex Gaussian data assumption are reviewed. Recent advances in RMT explore limiting behavior of eigenvalues and eigenvectors of random matrices as dimensions become increasingly large (referred to as infinite RMT). Defining the notion of an empirical distribution for the eigenvalues deviates from classical treatments, but yields amazing convergence toward deterministic distributions. Coupled with the Stieltjes transform, powerful tools emerge that can provide new insights especially for signal processing methods intimately tied to the eigen-decomposition, e.g. diagonal loading and dominant mode rejection used in adaptive beamforming. Although many of the theorems are based on asymptotic convergence as dimensionality increases, they describe performance of finite systems quite well. Aspects of infinite RMT are also reviewed and contrasted with classical RMT.

8:45

3aUW2. Constructing acoustic timefronts using random matrix theory. Katherine C. Hegewisch and Steven Tomsovic (Physics, Washington State University, Deparment of Physics, Pullman, WA 99164-2814, tomsovic@wsu.edu)

In a recent letter [Europhys.Lett. 97, 34002 (2012)], random matrix theory is introduced for long-range acoustic propagation in the ocean. The theory is expressed in terms of unitary propagation matrices that represent the scattering between acoustic modes due to sound speed fluctuations induced by the ocean's internal waves. The scattering exhibits a power-law decay as a function of the differences in mode numbers thereby generating a power-law, banded, random unitary matrix ensemble. This talk describes that approach and extends the methods to the construction of an ensemble of acoustic timefronts. The result is a very efficient method for studying the statistical properties of timefronts at various propagation ranges that agrees well with propagation based on the parabolic equation. It helps identify which information about the ocean environment survives in the timefronts and how to connect features of the data to the surviving environmental information. It also makes direct connections to methods used in other disordered wave guide contexts where the use of random matrix theory has a multi-decade history. [This work was supported by ONR and NSF.]

9:15

3aUW3. Random matrix theory and performance prediction of subspace methods. Raj R. Nadakuditi (University of Michigan, Ann Arbor, MI 48109, rajnrao@umich.edu)

Subspace methods constitute a powerful class of techniques for detection, estimation and classification of signals buried in noise. Recent results in random matrix theory precisely quantify the accuracy of subspaces estimates from finite, noisy data in both the white noise and colored noise setting. This advance facilitates unified performance analysis of signal processing methods that rely on these empirical subspaces. We discuss the pertinent theory and its application to the characterization of the performance of direction-of-arrival estimation, matched subspace detection and subspace clustering for large arrays in the sample-starved setting for both white and colored noise.

9:45

3aUW4. Cross-correlations of diffuse ocean noise using eigenvalue based statistical inference. Ravi Menon, Peter Gerstoft, and William S. Hodgkiss (SIO, 9500 Gilman Dr, UC San Diego, La Jolla, CA 92093, rmenon@ucsd.edu)

Cross-correlations of diffuse noise fields can be used to extract environmental information. The influence of directional sources (usually ships), often results in a bias of the travel time estimates obtained from the cross-correlations. Using an array of sensors, insights from random matrix theory on the behavior of the eigenvalues of the sample covariance matrix (SCM) in an isotropic noise field are used to isolate the diffuse noise component from the directional sources. A sequential hypothesis testing of the eigenvalues of the SCM reveals eigenvalues dominated by loud sources that are statistical outliers for the assumed diffuse noise model. Travel times obtained from cross-correlations using only the diffuse noise component (i.e., by discarding or attenuating the outliers) converge to the expected travel times (i.e., unbiased estimates) and are stable temporally. Data from the Shallow Water 2006 experiment demonstrates the effectiveness of this approach and that the SNR builds up as the square root of time, as predicted by theory.

10:00

3aUW5. Eigenvalues of the sample covariance matrix for a towed array. Peter Gerstoft, Ravishankar Menon, William S. Hodgkiss (SIO Marine Phys Lab, Univ of California San Diego, 9500 Gillman Drive, La Jolla, CA 92093-0238, gerstoft@ucsd.edu), and Christoph Mecklenbrauker (Vienna University of Technology, Vienna, Austria)

Observations of the spatial sample covariance matrix (SCM) reveal that the ordered noise eigenvalues of the SCM decay steadily. Using a stochastic model for the sample covariance matrix, the empirical eigenvalue distribution can be derived using random matrix theory. The eigenvalue spectrum is directly related to the quantile function of the empirical eigenvalue distribution. These ordered eigenvalues have a decay that resembles what is observed in real data. Noise on the array is considered either incoherent self-noise or propagating acoustic noise that is coherent across the array. Using conventional 2D or 3D isotropic noise models, realizations of the SCM eigenvalues are generated using random matrix theory. Deep-water towed-array data are analyzed and it is shown that the eigenvalues of the SCM and compares well with theory.

10:15

3aUW6. Random matrix theory analysis of the dominant mode rejection beamformer. Kathleen E. Wage (Electrical and Computer Engineering Dept., George Mason University, 4400 University Drive, MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu) and John R. Buck (Electrical and Computer Engineering Dept., University of Massachusetts Dartmouth, North Dartmouth, MA)

The Dominant Mode Rejection (DMR) beamformer developed by Abraham and Owsley [Proc. Oceans, 1990] determines the beamformer weights from the sensor covariance matrix eigendecomposition. The weights are designed to reject signals contained in the dominant subspace, which is defined by the eigenvectors associated with the largest eigenvalues. In previous work, we developed a model for the mean notch depth (ND) of the DMR beamformer from random matrix theory (RMT) results on the sample eigenvector fidelity [IEEE Stat. Sig. Proc. workshop, 2012]. While ND is useful, other metrics such as white noise gain (WNG) and signal to interference and noise ratio (SINR) are of great interest. WNG characterizes the beamformer robustness to mismatch, and SINR quantifies overall performance. SINR loss is defined as the ratio of the SINR for a beamformer designed using sample statistics to the SINR for the optimal beamformer designed using ensemble statistics. This talk extends our previous work by considering the relationship among ND, WNG, and SINR for the DMR beamformer. A surprising result obtained from RMT is that for a single loud interferer and twice as many snapshots as sensors, the expected SINR loss depends only on the number of snapshots. [Work supported by ONR.]

10:30

3aUW7. Computational model for the eigenvalue density function of a cylindrically isotropic noise sample covariance matrix. Saurav R. Tuladhar, John R. Buck (ECE Dept, University of Massachusetts Dartmouth, 285 Old Westport Rd, North Dartmouth, MA 02747, stuladhar@umassd.edu), and Kathleen E. Wage (ECE Dept, George Mason University, Fairfax, VA)

Adaptive beamformers (ABFs) rely on the knowledge of ensemble covariance matrix (ECM), which is usually not known a priori. The ECM is often estimated from the sample covariance matrix (SCM). When the sample size is limited, the SCM may not converge to the ECM. ABF performance is then determined by the SCM eigenstructure. Random Matrix Theory (RMT) provides a helpful framework to analyze the asymptotic behavior of the SCM eigenstructure. This talk presents a computational model for the SCM's asymptotic eigenvalue density function (EDF) for a uniform linear array in a cylindrically isotropic noise field. Cylindrically isotropic noise fields are common models for shallow water environments, including the Kuperman-Ingenito noise model. The proposed method employs Nadakuditi and Edelman's polynomial method to model the EDF as the product of a deterministic matrix and a Wishart matrix. The model exploits properties of free multiplicative convolution to reduce the required polynomial order, resulting a substantial computational savings. The model EDF exhibits good agreement with eigenvalue histograms from simulations. A model for the SCM EDF is a necessary first step for accurate bearing dependent detection thresholds in cylindrically isotropic noise environments. [Supported by ONR.]

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

A. T. Herfat, Chair ASC S2
Emerson Climate Technologies, Inc., 1675 West Campbell Road, PO Box 669, Sidney, OH 45365-0669

C. F. Gaumond, Vice Chair ASC S2
Naval Research Laboratory, Code 7142, 4555 Overlook Ave. SW, Washington, DC 20375-5320

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and four of its five subcommittees, take note—that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 23 October 2012.

Scope of S2: Standards, specifications, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance and comfort.

WEDNESDAY AFTERNOON, 24 OCTOBER 2012 JULIA LEE A/B, 1:00 P.M. TO 3:00 P.M.

Session 3pAB

Animal Bioacoustics: Vocalization, Hearing, and Response in Non-Human Vertebrates II

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

Contributed Papers

1:00

3pAB1. Aerial hearing sensitivity in a southern sea otter (*Enhydra lutris nereis*). Asila Ghoul and Colleen Reichmuth (Institute of Marine Sciences, Long Marine Laboratory, University of California, Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, asila@ucsc.edu)

The lack of information concerning auditory sensitivity in sea otters has been recognized by biologists and resource managers as a priority research need for this threatened species. Noise-generating human activity in near-shore marine environments occurs as a result of construction, transportation, exploration and recreation. These activities may degrade critical habitat or cause behavioral or auditory effects that are harmful to individuals. As direct measures of hearing are not presently available for sea otters, we obtained psychophysical hearing thresholds from a trained individual. Audiometric testing was conducted with an adult male sea otter using 500 ms frequency-modulated narrow-band sweeps under quiet conditions. Absolute aerial thresholds were collected at eleven frequencies ranging from 0.125 to 45 kHz. The sea otter showed a broad functional range of hearing, extending from 0.250 to ~40 kHz, with best sensitivity between 2 and 16 kHz. The lowest measured threshold was -1 dB re 20 μPa at 8 kHz. The high-frequency hearing data was similar to that of terrestrial carnivores, while hearing thresholds below 1 kHz showed a relative decrease in sensitivity. Measurements of underwater sensitivity in the same sea otter are ongoing, and will inform explorations of amphibious hearing capabilities in marine mammals, as well as provide insight into the effects of anthropogenic noise on this vulnerable species.

1:15

3pAB2. Auditory thresholds in marine vertebrates conform to natural ambient noise levels. Michael A. Stocker (Ocean Conservation Research, P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org) and John T. Reuterdahl (Ocean Conservation Research, Mill Valley, CA)

Auditory thresholds are often displayed in a manner that reveals what is commonly called a “U-curve.” But if the threshold curves are displayed on the x axis on a true Log10 scale the profile is shaped differently. For marine mammals the shape is more like a “hockey stick.” If these curves are overlaid on the “Wenz ambient noise spectra curves” there appears to be shape conformance. This makes sense as auditory sensitivity would naturally evolve to exclude ambient environmental noise. This paper evaluates 120 legacy auditory threshold curves from 18 species of marine mammals and 60 threshold curves from 32 species of fish. The auditory threshold curves from the fish do not conform to the Wenz curves. Given that both the auditory thresholds and the Wenz curves were expressed as pressure gradient energy it is possible that the profile of the fish threshold curves express

sound in either the particle velocity, or both particle velocity and pressure gradient energy. This paper extrapolates the particle velocity data from the fish threshold conditions to determine if there is some conformity to ambient noise levels in either or both the particle and pressure gradient realms.

1:30

3pAB3. High-frequency hearing in seals and sea lions and the implications for detection of ultrasonic coded transmitters. Kane A. Cunningham (Department of Ocean Sciences, University of California at Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, kacunningham413@yahoo.com), Sean A. Hayes (Fisheries Ecology Division, NOAA National Marine Fisheries Service, Southwest Fisheries Science Center, Santa Cruz, CA), Michelle W. Rub (Fish Ecology Division, NOAA National Marine Fisheries Service, Northwest Fisheries Science Center, Seattle, WA), and Colleen Reichmuth (Institute of Marine Sciences, Long Marine Laboratory, University of California, Santa Cruz, CA)

In order to better understand the ability of pinnipeds to detect acoustic signals from ultrasonic coded transmitters (UCTs) commonly used in fisheries research, high-frequency hearing thresholds were obtained from a trained Pacific harbor seal (*Phoca vitulina*) and a trained California sea lion (*Zalophus californianus*). Using a 69 kHz, 500 ms, narrow-band FM sweep stimulus, detection thresholds for the harbor seal and the sea lion were determined to be 106 dB and 112 dB re 1 μPa respectively. While the harbor seal threshold falls within the range of existing data, the sea lion threshold is 33 dB lower than expected based on previous reports. This finding indicates that sea lions may be more sensitive to the output of UCTs than previously thought, and allows for the possibility that acoustically tagged fish may be selectively targeted for predation by sea lions as well as seals. These hearing thresholds, combined with ongoing work on the effect of signal duration on high-frequency hearing, will help estimate the ranges at which certain UCTs can be detected by these species. Detection range estimations, in turn, will allow fisheries researchers to better understand how fish survivorship data obtained using UCTs may be skewed by pinniped predation.

1:45

3pAB4. Animal-borne active acoustic tags: A new paradigm to conduct minimally invasive behavioral response studies? Holger Klinck (Cooperative Institute for Marine Resources Studies, Oregon State University and NOAA Pacific Marine Environmental Laboratory, Hatfield Marine Science Center, 2030 SE Marine Science Drive, Newport, OR 97365, Holger.Klinck@oregonstate.edu), Markus Horning, David K. Mellinger (Oregon State University, Newport, OR), Daniel P. Costa (University of California, Santa Cruz, CA), Selene Fregosi (Oregon State University, Newport, OR), David A. Mann (Loggerhead Instruments, Sarasota, FL), Kenneth Sexton (The Sexton Company, Salem, OR), and Luis Huckstadt (University of California, Santa Cruz, CA)

In 2011 a pilot study was begun to evaluate the potential of animal-borne active acoustic tags for conducting minimally-invasive behavioral response studies on pinnipeds. A basic prototype tag was developed and tested on juvenile northern elephant seals (Mirounga angustirostris) during translocation experiments at Año Nuevo State Park, CA, USA in spring 2012. The principal scientific questions of this pilot study were these: (1) do sounds emitted from an animal-borne low acoustic intensity tag elicit behavioral responses, and (2) are potential animal responses related to signal content (e.g., threatening vs. non-threatening). Although the sample size was small, preliminary results indicate that (1) low-intensity sounds emitted by animal-borne tags elicit distinct behavioral responses, (2) these responses appear related to signal content, and (3) the responses may differ based on depth, bathymetry, and location. The results of the conducted study show the promise of this approach as a minimally-invasive and cost-effective method to investigate animal responses to underwater sounds, as well as a method to develop mitigation strategies. Future efforts would increase the sample size, range of acoustic stimuli, and age/sex classes of tagged seals. [Funding from NOAA/NMFS Ocean Acoustics Program.]

2:00

3pAB5. Tracking calling depths and movements of North Atlantic right whales using multipath localization. Robert D. Valtierra (Mech. Engineering, Boston University, 110 Cummington St., Boston, MA 02215, rvaltier@bu.edu), Sofie M. VanParijs (Northeast Fisheries Science Center, National Oceanic and Atmospheric Administration, Woods Hole, MA), R. G. Holt (Mech. Engineering, Boston University, Boston, MA), and Danielle M. Cholewiak (Northeast Fisheries Science Center, National Oceanic and Atmospheric Administration, Woods Hole, MA)

The track and calling depths of a North Atlantic right whale (NARW) recorded by 10 bottom-mounted Autonomous Acoustic Recording Units (ARUs) in the Stellwagen Bank National Marine Sanctuary was determined using the Direct Reflected Time Difference of Arrival (DRTD) localization method. An autocorrelation technique was used to extract direct-reflected time difference of arrival information from recorded NARW up-calls containing several overlapping multipath signal arrivals. The method's feasibility was tested using data from play back transmissions to localize an acoustic transducer at a known depth and location. The method was then used to track an hour of movements and depths of a single NARW using periodic up-calls for localization purposes.

2:15

3pAB6. Passive acoustic monitoring on the North Atlantic right whale calving grounds. Melissa Soldevilla, Lance Garrison (NOAA-NMFS Southeast Fisheries Science Center, 75 Virginia Beach Dr., Miami, FL 33149, melissa.soldevilla@noaa.gov), and Christopher Clark (Bioacoustics Research Program, Cornell University, Ithica, NY)

Shallow water environments, such as the North Atlantic right whale calving grounds, pose a challenge to cetacean passive acoustic monitoring due to high variability in ambient noise and environmental conditions. In this region of high shipping traffic and increased ship-strike risk, passive acoustic monitoring may reduce right whale ship strikes. This study describes temporal variability in right whale call detections, ambient noise sources, and environmental conditions on the right whale calving grounds during 2009-2010 and 2010-2011. Right whale detections occurred between November 19 and March 11, on up to 25% of days per deployment with increased nocturnal call detections, and increased acoustic presence off Jacksonville, FL during 2010-2011. Shipping noise was most common off Jacksonville, detected in up to 74% of minutes, with a diurnal peak, while tidally-associated broadband impulses were detected in up to 43% of minutes off Savannah GA. Environmental conditions including SST, wind, waves, and tidal height varied on daily and semi-diurnal scales. While sightings were higher in 2009-2010, the fewer sightings in 2010-2011 were more narrowly distributed within the depth range of the acoustic instruments. Passive acoustic monitoring is effective for detecting right whales in this environment, especially at night when they cannot be seen.

2:30

3pAB7. Comparison of the first-year response of beaked and sperm whale populations to the Northern Gulf oil spill based on passive acoustic monitoring. Natalia Sidorovskaia (Physics, Univ. of Louisiana, P.O. Box 44210, Lafayette, LA 70504-4210, nas@louisiana.edu), Azmy Ackleh (Mathematics, Univ. of Louisiana, Lafayette, LA), Christopher Tiemann (Applied Research Laboratories, UT Austin, Austin, TX), Juliette Ioup, and George Ioup (Physics, Univ of New Orleans, New Orleans, LA)

This paper continues a discussion on using passive acoustic methods to study the environmental impact of the recent oil spill in the Northern Gulf of Mexico on resident populations of marine mammals. The Littoral Acoustic Demonstration Center, possessing several broadband acoustic datasets collected near the spill site before and after the event, is in a unique position for monitoring long-term environmental impacts in the vicinity of the incident The pre-spill recordings provide a baseline which, when combined with post-spill measurements, give important indicators of changes in the local populations. Ackleh et al., J. Acoust. Soc. Am. 131, 2306-2314, provide a comparison of 2007 and 2010 measurements showing a decrease in acoustic activity and abundance of sperm whales at the 9-mile distant site, whereas acoustic activity and abundance at the 25-mile distantsite has clearly increased. This may indicate that some sperm whales have relocated farther away from the spill subject to food source availability. This paper reports on applying

developed population estimation techniques to monitor beaked whale response that appears to be different from that of sperm whales. Follow-up experiments will be critical for understanding the long-term impact on different species. [Research supported by SPAWAR, NSF, and Greenpeace.]

2:45

3pAB8. Population density of sperm whales in the Bahamas estimated using non-linked sensors. Elizabeth T. Küsel (Northwest Electromagnetics and Acoustics Research Laboratory, Portland State University, 1900 SW 4th Ave., Portland, OR 97201, kusele@alum.rpi.edu), David K. Mellinger (Cooperative Institute for Marine Resources Studies, Oregon State University and NOAA Pacific Marine Environmental Laboratory, Newport, OR), Len Thomas (Centre for Research into Ecological and Environmental Modelling, University of St. Andrews, St. Andrews, Fife, United Kingdom), and Tiago A. Marques (Centre for Research into Ecological and Environmental Modelling, University of St. Andrews, Campo Grande, Lisboa, Portugal)

Estimates are presented of sperm whale click detection probability and sperm whale population density at the U.S. Navy's Atlantic Undersea Test and Evaluation Center (AUTEC) in the Bahamas. The estimation of the probability of detecting whale echolocation clicks at multiple non-linked sensors uses estimates of sperm whale source level distribution, beam pattern of click emission, distribution of whale locations and orientations with respect to the sensors while clicking, acoustic transmission loss from source (whale) to receiver (bottom hydrophone), and noise levels at the receiver. These data are combined in a Monte Carlo model that propagates simulated clicks from whales at various random positions to each receiving hydrophone to estimate the signal-to-noise ratio at the receiver and the detection function, the probability of detecting clicks as a function of distance. The estimated detection function for each receiving hydrophone is then combined with information on the detector's rate of missed calls and false detections as a function of signal-to-noise ratio, average sperm whale click rates, and the actual number of clicks detected in a given period of time in order to estimate population density. Results are compared to multi-sensor cases where detection functions were estimated analytically.

WEDNESDAY AFTERNOON, 24 OCTOBER 2012 — TRUMAN A/B, 1:00 P.M. TO 3:00 P.M.

Session 3pBA

Biomedical Acoustics: Best Student Paper Award Poster Session

Kevin J. Haworth, Chair
University of Cincinnati, Cincinnati, OH 45209

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with $500 for first prize, $300 for second prize, and $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee. Below is a list of students competing, with their abstract numbers and titles listed. Full abstracts can be found in the oral sessions associated with the abstract numbers.

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

2aBA6. Modeling acousto-optic sensing of high-intensity focused ultrasound lesion formation. Student author: Matthew T. Adams

2pBA14. Compound manipulation of micro-particles using a single device: Ultrasonic trapping, transporting and rotating. Student author: Kun Jia

3aBA7. Focused, radially-polarized shear wave beams in tissue-like media. Student author: Kyle S. Spratt

3aBA8. Shear wave generation using hybrid beamforming methods. Student author: Alireza Nabavizadeh

3aBA13. Rayleigh wave propagation method for the characterization of viscoelastic properties of biomaterials. Student author: Siavash Kazemirad

4aBA1. Effects of encapsulation damping on frequency dependent subharmonic threshold for contrast microbubbles. Student author: Amit Katiyar

4aBA2. Pulse duration dependence of cavitation emissions and loss of echogenicity from ultrasound contrast agents insonified by Doppler pulses. Student author: Kirthi Radhakrishnan

4aBA3. Echogenicity and release characteristics of folate-conjugated echogenic liposomes for cytosolic delivery of cancer drugs. Student author: Shirshendu Paul

4aBA4. High-frequency harmonic imaging with coded excitation: Implications for the assessment of coronary atherosclerosis. Student author: Himanshu Shekhar

4aBA6. Acoustic emissions associated with ultrasound-induced rupture of *ex vivo* blood vessels. Student author: Cameron L. Hoerig

4aBA7. Cavitation mechanisms in ultrasound-enhanced permeability of *ex vivo* porcine skin. Student author: Kyle T. Rich

4aBA8. Laser-induced-cavitation enhanced ultrasound thrombolysis. Student author: Huizhong Cui

4aBA9. Ethanol injection induced cavitation and heating in tissue exposed to high intensity focused ultrasound. Student author: Chong Chen

4pBA1. Effect of skull anatomy on intracranial acoustic fields for ultrasound-enhanced thrombolysis. Student author: Joseph J. Korfhagen

4pBA6. Histological analysis of biological tissues using high-frequency ultrasound. Student author: Kristina M. Sorensen

4pBA8. Parametric imaging of three-dimensional engineered tissue constructs using high-frequency ultrasound. Student author: Karla P. Mercado

WEDNESDAY AFTERNOON, 24 OCTOBER 2012 BASIE A1, 1:30 P.M. TO 2:20 P.M.

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Preston S. Wilson, Chair
Applied Research Lab., Univ. of Texas at Austin, Austin, TX 78712-0292

Chair's Introduction—1:30

Invited Paper

1:35

3pED1. Physclips: Multimedia, multi-level learning, and teaching resources. Joe Wolfe and George Hatsidimitris (University of New South Wales, School of Physics, University of New South Wales, Sydney, NSW 2052, Australia, j.wolfe@unsw.edu.au)

Physclips provides multimedia resources to physics students and teachers at the levels of senior high school to introductory university. Completed volumes cover mechanics, waves and sound. Each chapter includes a rich multimedia lesson of about 10 minutes, including film clips, animations, sound files and images of key experiments and demonstrations. Contextually embedded links lead to html pages providing broader and deeper support and, where needed, to tools such as calculus and vectors. The ongoing development of the interface reflects learner feedback and our own experience and research. The architecture and presentation of Physclips is largely consistent with evidence-based guidelines in the field of educational multimedia. Often, animations and labeling are superimposed on film clips to indicate abstract quantities, thus providing the novice with the insight of the expert's 'mind's eye'. The scrollbar is indexed with keywords and images to assist learners to find and to relocate conceptually discrete segments, which facilitates revision and reference usage. Together with extensive cross-linking, this allows students to construct individual learning pathways. Teachers download animations singly or in compressed folders for inclusion in lessons, blogs etc. Physclips is supported by Australia's Office of Learning and Teaching and the University of New South Wales.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Lily M. Wang, Chair
Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, Omaha, NE 68182-0816

Chair's Introduction—1:30

Invited Papers

1:35

3pID1. Hot topics in speech communication: Listening to foreign-accented speech. Tessa Bent (Department of Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

There are currently more non-native English speakers in the world than there are native speakers. Most of these second language users will speak with a detectable foreign accent. Foreign-accented speech differs from native language norms along many acoustic-phonetic dimensions including the realization of vowel, consonant, and prosodic features. An important question for researchers in the field of speech communication is how this type of language variation influences speech perception and perceptual processing. Numerous findings have shown that foreign-accented speech is generally less intelligible, receives lower comprehensibility ratings, and is processed more slowly than native-accented speech. Further, these negative perceptual effects can be exacerbated by noisy listening conditions or listener variables such as age or hearing loss. However, research over the past several years has shown the amazing flexibility of the speech perception mechanism in its ability to adapt to this form of variability. Through experience and training, listeners can improve their word identification skills with specific foreign-accented talkers, particular foreign accents, and foreign-accented speech in general. New directions in this research area include perception of foreign-accented speech by infants and children as well as how a foreign accent may influence memory.

2:00

3pID2. New directions for manipulation of sound using acoustic metamaterials. Christina J. Naify, Gregory J. Orris, Theodore P. Martin, and Christopher N. Layman (Code 7160, Naval Research Laboratory, 4555 Overlook Ave SW, Washington, DC 20375, christina.naify.ctr@nrl.navy.mil)

Manipulation of sound waves using acoustic metamaterials has expanded significantly in recent years. Acoustic metamaterials are a class of materials that use sub-wavelength structures to achieve effective bulk properties under acoustic excitation. Unusual effective physical properties, including negative bulk modulus, negative mass density, and negative index have been achieved using metamaterials. Additionally, the development of structures based on transformational acoustics has resulted in designs for scattering reduction and sound focusing. Current research emphases include expansion from narrowband, resonant structures to broadband structures, as well as the design and construction challenges of three-dimensional structures. Active or tunable structures are also being explored. Examples will be given of negative index and three-dimensional metamaterial structures. [Work is supported by the Office of Naval Research.]

2:25

3pID3. Photoacoustic tomography: Ultrasonically breaking through the optical diffusion limit. Lihong Wang (Department of Biomedical Engineering, Washington University, One Brookings Drive, P.O. Box 1097, St. Louis, MO 63130-4899, lhwang@biomed.wustl.edu)

Photoacoustic tomography (PAT), combining optical and ultrasonic waves via the photoacoustic effect, provides in vivo multiscale non-ionizing functional and molecular imaging. Light offers rich tissue contrast but does not penetrate biological tissue in straight paths as x-rays do. Consequently, high-resolution pure optical imaging (e.g., confocal microscopy, two-photon microscopy, and optical coherence tomography) is limited to depths within the optical diffusion limit (~1 mm in the skin). In PAT, pulsed laser light penetrates the tissue and generates a small but rapid temperature rise, which induces emission of ultrasonic waves due to thermoelastic expansion. The ultrasonic waves, ~1000 times less scattering than optical waves in tissue, are then detected to form high-resolution images at depths up to 7 cm, breaking through the optical diffusion limit. PAT is the only modality capable of imaging across the length scales of organelles, cells, tissues, and organs with consistent contrast. Such a technology has the potential to enable multiscale systems biology and accelerate translation from microscopic laboratory discoveries to macroscopic clinical practice. PAT may also hold the key to the earliest detection of cancer by in vivo label-free quantification of hypermetabolism, the quintessential hallmark of cancer. The technology is commercialized by several companies.

Session 3pNS

Noise, ASA Committee on Standards, and Psychological and Physiological Acoustics: Passive and Active Noise Reduction in Hearing Protection

Richard L. McKinley, Cochair
Air Force Research Lab., Wright-Patterson AFB, OH 45433-7901

Hilary L. Gallagher, Cochair
Air Force Research Lab., Wright-Patterson AFB, OH 45433-7901

Chair's Introduction—1:00

Invited Papers

1:05

3pNS1. Development of an advanced hearing protection evaluation system. Kevin Shank and Josiah Oliver (Adaptive Technologies Inc., 2020 Kraft Dr Ste 3040, Blacksburg, VA 24060, kevin@adaptivetechinc.com)

Acoustic Test Fixtures (ATFs) are practical and often necessary tools for testing Hearing Protection Devices (HPDs) especially with extremely loud impulsive and/or continuous noise, for which the use of live subjects might not be advisable. Although there have been various standardized and laboratory ATFs from past research, there still exists large uncertainty in the correlation between the attenuation results obtained from ATFs and those obtained from actual human subject tests, particularly for intraaural HPDs. It is suspected that one of the main factors contributing to the discrepancy may be insufficient fidelity in the circumaural/intraaural flesh and bone. Human subject testing was performed to obtain median parameters of ear canal geometry and eardrum reflectance, which are considered to be critical parameters for circumaural/intraaural HPD attenuation performance. This presentation discusses the research methodologies and design implementation of these important subsystems in this advanced Hearing Protection Evaluation System (HPES).

1:25

3pNS2. Two case studies for fit testing hearing protector devices. William J. Murphy, Christa L. Themann, Mark R. Stephenson, and David C. Byrne (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

Hearing protection devices (HPDs) are typically selected based upon the Noise Reduction Rating (NRR) and, until recently, were rarely tested for attenuation in real-world environments. The National Institute for Occupational Safety and Health has developed a fit-testing system (HPD Well-Fit^{TM}) that performs attenuation tests with a large circumaural earmuff, a portable computer and a computer mouse with a scroll wheel. HPD Well-Fit was used to estimate the attenuation of employees working in two different settings: inspectors for off-shore drilling rigs and sandblasters at a hydroelectric facility. The highest exposure levels for the inspectors and sandblasters were estimated to be 110 and 130 dBA, respectively. Fit testing and training were used to achieve a 25-dB Personal Attenuation Rating (PAR) for the inspectors. Fit testing before and after the sandblaster work shift demonstrated PARs of 30 to 42 dB using HPD Well-Fit. The average time to complete the fit tests was 10 minutes. If retraining was necessary, then an additional 3 to 6 minutes were required.

1:45

3pNS3. Continuous and impulsive noise attenuation performance of passive level dependent earplugs. Richard L. McKinley, Hilary L. Gallagher (Air Force Research Laboratory, 2610 Seventh Street, Wright Patterson AFB, OH 45433, richard.mckinley@wpafb.af.mil), Melissa Theis (Oak Ridge Institute for Science and Education, Dayton, Ohio), and William J. Murphy (National Institute for Occupational Safety and Health, Cincinnati, OH)

Level dependent hearing protectors, earplugs and earmuffs, have advanced in technology due to the needs of military personnel and others to reduce the risk of hearing damage from impulsive noise. These hearing protectors were developed to preserve ambient listening capabilities therefore improving situational awareness while reducing the risk of noise induced hearing loss by attenuating both continuous and impulsive noise. Four commercially available passive level dependent earplugs were assessed for both continuous noise attenuation and impulsive insertion loss performance. The continuous noise attenuation results were collected using American National Standard Institute (ANSI) S12.6-2008 Methods for Measuring the Real-Ear Attenuation of Hearing Protectors while the impulsive insertion loss results were collected using ANSI S12.42-2010 Methods for the Measurement of Insertion Loss of Hearing Protection Devices in Continuous or Impulsive Noise Using Microphone-in-Real-Ear (MIRE) or Acoustic Test Fixture Procedures. The presentation will include the passive noise attenuation performance of level dependent earplugs for both continuous and impulsive noise. The impulsive insertion loss results for these particular hearing protectors will be applied to impulsive noise damage risk criteria for an estimate of allowable impulsive noise exposure.

2:05

3pNS4. Effective attenuation performance of passive hearing protectors: A temporary threshold shift study. Richard L. McKinley, Hilary L. Gallagher (Air Force Research Laboratory, 2610 Seventh Street, Wright Patt AFB, OH 45433, richard.mckinley@wpafb.af.mil), and Melissa Theis (Oak Ridge Institute for Science and Technology, Dayton, Ohio)

Passive hearing protectors have been used for decades to reduce the risk of noise induced hearing loss. Hearing protectors (earmuffs, earplugs, helmets) have traditionally been the first line of defense for personnel working in hazardous noise environments. According to ANSI S12.68-2007, the "gold standard" method of estimating effective A-weighted sound pressure levels when hearing protectors are worn is the classical octave band method. The octave band method subtracts the hearing protector noise attenuation from the ambient noise level for each relevant octave band to estimate the noise exposure at the ear, under the hearing protector. ANSI S12.6-2008 Methods for Measuring the Real-Ear Attenuation of Hearing Protectors was used to measure the attenuation of the hearing protectors. The purpose of this study was to measure the effective attenuation of a hearing protector in terms of temporary threshold shift (TTS) response for individual human subjects with and without hearing protection. This presentation will include the TTS response curves for subjects exposed to various noise levels and durations in a controlled laboratory environment. The passive hearing protectors evaluated in this study included an earplug, earmuff, and a headphone with minimal attenuation as determined by REAT.

Contributed Papers

2:25

3pNS5. Measurements of bone-conducted impulse noise from weapons using a head simulator. Odile H. Clavier, Anthony J. Dietz, Jed C. Wilbur (Creare Inc., 16 Great Hollow Rd, Hanover, NH 03755, ohc@creare.com), Edward L. Zechmann, and William J. Murphy (Hearing Loss Prevention Team, National Institute for Occupational Safety and Health, Cincinnati, OH)

High-intensity impulse sounds are generally considered to be more damaging than continuous sounds, so understanding the attenuation performance of hearing protection devices against impulse noise is key to providing adequate protection for exposed persons. The maximum attenuation of hearing protection devices is limited by bone-conducted sound. Weapon fire noise in the form of short duration impulses can reach peak levels of 170 dB SPL at the shooter's ear, a sound level for which maximum hearing protection is recommended and for which bone-conducted sound will be a significant factor. However, current acoustic test fixtures do not capture the bone-conducted sound paths. In this study, an anatomically correct head simulator built specifically to measure bone-conducted sound was used to evaluate the effects of impulse noise generated by hand guns and rifles at several peak sound pressure levels ranging between 120 dB SPL and 170 dB SPL. Time histories of the acceleration of the temporal bones and the sound pressure transmitted into the cranial cavity were recorded. Results investigating the linearity of the bone-conducted response to impulse noise at high peak levels and the effects of hearing protection on the sound level and vibrations inside the head are presented.

2:40

3pNS6. Adaptive feedforward control for active noise cancellation in-ear headphones. Sylvia Priese, Christoph Bruhnken (Institute of Measurement and Automatic Control, Leibniz Universität Hannover, Nienburger Straße 17, Hannover, 30167, Germany, sylvia.priese@imr.uni-hannover.de), Daniel Voss, Jürgen Peissig (Technology and Innovation, Sennheiser Electronic GmbH & Co. KG, Wedemark, NI, Germany), and Eduard Reithmeier (Institute of Measurement and Automatic Control, Leibniz Universität Hannover, Hannover, NI, Germany)

Noise can be disturbing, stressful or even harmful. Headphones with active noise cancellation (ANC) can enhance the user's comfort, especially when travelling. On a plane or a train, in the street or at work, these headphones give the possibility to reduce unwanted noise. The range of ANC headphones on the market is constantly increasing. Circumaural and supra-aural headphones with different control strategies have been available for a long time; over the last few years the product lines have been expanded to in-ear headphones. These headphones already have quite a good passive attenuation and are equipped with feedforward control for active noise cancellation. The best results in attenuation are achieved by semi-adaptive digital controls, which choose the best filter depending on the noise spectrum and can be manually adapted to the user. A fully adaptive control has already been proven to be very effective in aviation headsets and other ANC applications. Besides the market analysis of ANC headphones we would like to present an adaptive feedforward control for in-ear headphones and highlight the advantages compared to a static feedforward control.

2:55

3pNS7. Design of a feedback controller for active noise control with in-ear headphones. Christoph Bruhnken, Sylvia Priese (Institute of Measurement and Automatic Control, Leibniz Universität Hannover, Nienburger Straße 17, Hannover, 30167, Germany, christoph.bruhnken@imr.uni-hannover.de), Hatem Foudhaili, Jürgen Peissig (Technology and Innovation, Sennheiser Electronic GmbH & Co. KG, Wedemark, NI, Germany), and Eduard Reithmeier (Institute of Measurement and Automatic Control, Leibniz Universität Hannover, Hannover, NI, Germany)

Nowadays mobility is an important factor in many jobs. Therefore, there is an increased use of planes, trains and cars, and the associated exposure to noise. Good acoustic insulation is often hard to realize due to the involved extra weight. Ear protection or headphones with active noise control (ANC) may be a possible solution. Today circumaural and supra-aural ANC headphones with good attenuation are commercially available. However, their weight and the necessary headband can impair the wearing comfort. ANC in-ear headphones do not have these disadvantages and, therefore, there is a need of further research in the field of ANC. In ANC headphones, disturbing noise is minimized by an out-of-phase anti-noise. Therefore, the noise is recorded by microphones next to each ear, and filtered by an analog or digital platform to generate the anti-noise. There are two main control strategies depending on the position of the microphones, feedforward control with an external reference microphone and feedback control with an internal error microphone. The presentation will focus on the design of feedback controllers and the main problem regarded to in-ear headphones, interpersonal variances, which make the design of stable controllers with high noise attenuation difficult. A model-based solution will be presented.

Session 3pUW

Underwater Acoustics and Signal Processing in Acoustics: Advances in Underwater Acoustic Communication

Hee-Chun Song, Chair
Scripps Institution of Oceanography, La Jolla, CA 92093-0238

Contributed Papers

1:15

3pUW1. Active average intensity based on single vector sensor. Pengyu Du, Xiao Zhang, Jingwei Yin, and Xiao Han (College of Underwater Acoustic Engineering, Harbin Engineering University, Harbin, Heilongjiang 150001, China, yinjingwei@hrbeu.edu.cn)

Code divided multiple access underwater communication based on single vector sensor is studied in this paper. The most common methods to estimate azimuth with self-directivity of single vector sensor are average sound intensity method and complex sound intensity method, however, for the same frequency band multi-users, theoretical limitation for these methods is only two users.Spread spectrum communication is featured with strong anti-multipath, anti-interference, secret-keeping and communication web composing ability. Active average intensity method, which measures azimuths of multi-users simultaneously with the excellent correlative characteristics of pseudo-random code in spread spectrum communication, is proposed in this paper. Simulation and experiment for same frequency band spread spectrum multi-user communication testify the feasibility and utility of active average sound intensity method. With the estimated azimuth, vector combination can be generated to adjust the directivity of vector sensor, achieve multi-user beam communication, inhibit multi-path interference, and enhance processing gain and lower error rate. Key words: underwater acoustic communication; CDMA; single vector sensor; active acoustic intensity average

1:30

3pUW2. The application of differential spread spectrum technology in underwater acoustic communication. Xiao Han, Jingwei Yin, Pengyu Du, and Xiaoyu Guo (College of Underwater Acoustic Engineering, Harbin Engineering University, Mudanjiang, Harbin, Heilongjiang 150001, China, yinjingwei@hrbeu.edu.cn)

In underwater acoustic channel, the Doppler effect produced by relative movement between Source and information destination is very complex. Currently, the spread spectrum system typically uses PSK modulation. As the transmission characteristic in water sound channel is phase rapid changing, spread spectrum systems based on PSK modulation need high precision in estimating the carrier and need continuous tracking of the carrier, which make the performance in practical applications limited. Differential spread spectrum acoustic communication technology is studied in this paper. Using differential coherent demodulation method at the receiving end, which solves the problem of estimating the carrier in underwater acoustic communication, can overcome the frequency and phase error due to the drift of the carrier in transfer process. This method is verified Through computer simulation studies and Lake test.

1:45

3pUW3. Research on multilevel differential amplitude and phase-shift keying in convolution-coded orthogonal frequency division multiplexing underwater communication system. Yuheng Zhang, Chi Wang, Jingwei Yin, and Xueli Sheng (College of Underwater Acoustic Engineering, Harbin Engineering University, Harbin, Heilongjiang 150001, China, yinjingwei@hrbeu.edu.cn)

With the increasing demands of underwater source development and the increase of users, underwater transfer information has also greatly increased and high-bit-rate underwater acoustic communication has become a hot topic in underwater acoustic communication research. MDAPSK (Multilevel Differential Amplitude and Phase-shift Keying) is a modulation technique having high efficiency in spectrum utilization, which transfers information by using differential amplitude and phase code, demodulates information by adopting coherent demodulation. It reduces the difficulty of system, and improves the speed of transmission. While OFDM (Orthogonal Frequency Division Multiplexing) has advantages including high efficiency in spectrum and faster communication speed. After describing the schemes of the two technologies, a design scheme which concerns application of MDAPSK in the OFDM based underwater communication was given. The convolutional codes are also used in this system to realize the effectiveness and reliability in high-bit-rate underwater wireless communication. The computer simulation and the channel pool experimentation show that the system has a better performance. Key words: MDAPSK; OFDM; convolution coding; high-bit-rate communication

2:00

3pUW4. Application of orthogonal frequency division multiplexing in cognitive underwater communication. Chi Wang, Jingwei Yin, Pengyu Du, and Longxiang Guo (College of Underwater Acoustic Engineering, Harbin Engineering University, Mudanjiang, Harbin, Heilongjiang 150001, China, yinjingwei@hrbeu.edu.cn)

With the development of underwater acoustic communication in military and commercial field and the urgent need for underwater wireless Ad Hoc networks, developing an intelligent and high-bit-rate underwater communication system is imminent. OFDM(Orthogonal Frequency Division Multiplexing) technology could be a good platform for cognitive underwater communication, which has advantages including high efficiency in spectrum utilization, faster communication speed and flexibility in choosing frequencies. A design scheme of the OFDM based cognitive underwater communication and block diagram are given. The system can intelligently choose NC-OFDM (Non-Contiguous OFDM), DOFDM (Differential OFDM) or Pilot-added OFDM communication schemes in order to meet different channel conditions and different rate requirements and to overcome the problem of data conflict and the waste of spectrum resources in multi-users' competitive communication. Meanwhile, the system also can intelligently choose parameters in each scheme, such as sub-channel, pilot interval and error-correcting codes. The simulation results prove the feasibility and effectiveness of the OFDM based cognitive underwater communication.

2:15

3pUW5. Volterra series-based non-linearity analysis of shallow water acoustic channels. Xiaopeng Huang (Dept. of Electrical and Computer Engineering, Stevens Institute of Technology, Castle Point on Hudson, Hoboken, NJ 07030, xiaopeng.huang0508@gmail.com)

Most of existing underwater acoustic (UWA) communication systems are based on the linear UWA channels. However, some environmental factors (e.g., bubble plumes) in complicated shallow water environments will contribute to the non-linearity of channels. Therefore, In order to fully understand the properties of shallow water acoustic channels, and develop more bandwidth-efficient communication systems in complicated shallow

3p WED. PM

water environments, we adopt the Volterra series to analyze the non-linearity of shallow water acoustic channels for the first time, and its completed theoretical derivations will be presented. Volterra series combines the representations of a nonlinear system without memory and a linear, casual system with memory to describe a nonlinear system with memory. Generally speaking, the central problem in using a Volterra approach to the analysis of nonlinear channels with momory consists of estimating the Volterra kernels, which represent a nonparametric characterization of the channel.

2:30

3pUW6. Shallow water acoustic channel modeling with adaptive communications. Xiaopeng Huang (Dept. of Electrical and Computer Engineering, Stevens Institute of Technology, Castle Point on Hudson, Hoboken, NJ 07030, xiaopeng.huang0508@gmail.com)

The underwater acoustic channel is known to be severely bandwidth limited due to sound attenuation by sea water, and interaction with the ocean surface and bottom. Yet, shallow water acoustic channels at high frequencies are little understood, particularly in shallow water environments, and hence the quest for achieving a viable adaptive communication solution has been a challenge that perplexed scientists for a long time. In this abstract, we first take Hodson River estuary as an example to investigate the characterizations of shallow water environments, which mainly comprises the evaluation of key channel parameters such as the scattering function, Doppler shift, coherent bandwidth, coherent time, 2D (i.e., Time-Frequency) time-variant channel impulse response (CIR). The study will also cover channel fading statistics, and water conditions that affect the CIR (e.g., bubble plumes and mddium inhomogeneities). Finally, the models developed will be used to evaluate the achievable performance of channel estimation and adaptive communication systems in shallow water acoustic media.

2:45

3pUW7. Bidirectional equalization for underwater acoustic communications. Hee-Chun Song (Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu)

The bi-directional decision feedback equalizer (BiDFE) that combines the outputs of a conventional DFE and backward DFE can improve the performance of the conventional DFE by up to 1-2 dB based on simulations. In this paper, the BiDFE concept is extended to multi-channel time reversal communications involving a DFE as a post-processor. Experimental data collected in shallow water (10-20 kHz) show that the performance can be enhanced by 0.4-1.8 dB in terms of output SNR. In particular, a larger improvement (e.g., 1.8 dB) is achieved for time-varying channels where the channel diversity in opposite directions is more profound.

Plenary Session, Business Meeting, and Awards Ceremony

David L. Bradley, President
Acoustical Society of America

Business Meeting

Presentation of Certificates to New Fellows

Peter F. Assmann
Yang-Hann Kim
David A. Eddins
William J. Murphy
John A. Hildebrand
Scott D. Pfeiffer
Peter Howell
John R. Preston
Andrew J. Hull
Ronald C. Scherer

Presentation of Awards

Medwin Prize in Acoustical Oceanography to John A. Colosi

Rossing Prize in Acoustics Education to Joe Wolfe

Silver Medal in Animal Bioacoustics to Richard R. Fay

Silver Medal in Noise to Keith Attenborough

Silver Medal in Physical Acoustics to Andrea Prosperetti

Session 3eED

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

Marcia J. Isakson, Cochair
Applied Research Laboratories, University of Texas at Austin, Austin, TX 78713

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

Contributed Paper

5:30

3eED1. Hands-on demonstrations for Project Listen Up: Education outreach Part IV. Jacqueline A. Blackburn and Murray S. Korman (Physics Department, U.S. Naval Academy, Chauvenet Hall Room 295, Annapolis, MD 21402, korman@usna.edu)

Acoustical demonstrations geared to promote a hands-on learning experience for middle- and high-school age Girl Scouts are setup. The participants will be free to explore, control the apparatus and make their own scientific discoveries. The hands-on demonstrations will include (1) a homemade electric slide guitar using easy to find parts, (2) a safe smoke ring generator and (3) a portable ripple tank with plastic eyedroppers for simple excitation of waves.

WEDNESDAY EVENING, 24 OCTOBER 2012 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees on the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings 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	Mary Lou Williams
Signal Processing in Acoustics	Lester Young A

Session 4aAAa

Architectural Acoustics: The Technical Committee on Architectural Acoustics Vern O. Knudsen Distinguished Lecture

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

William J. Cavanaugh, Cochair
Cavanaugh Tocci Assoc Inc, 327F Boston Post Rd., Sudbury, MA 01776

Chair's Introduction—8:55

Invited Paper

9:00

4aAAa1. The consultant's risk is an invitation to academia—An exploration of the greatest successes in a design career thus far, and the research-based foundations that made them possible. Scott D. Pfeiffer (Threshold Acoustics LLC, 53 West Jackson Blvd., Suite 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

The path of discovery is common for both the academic and the consultant. The point of departure for the consultant is the decision which must be based on limited information. The resulting void in knowledge often invites research within the narrow scope necessary for academic certainty. Exploration of some past successes provides a roadmap for a closer relationship between academia and the consulting community. There are seminal papers that are universally used in bracketing design decisions. The strength of these is in their certainty, and the certainty comes from clear assumptions and limitations to the conditions of the studies. Too often, the consulting world levies criticism against ivory tower academia for these very limitations, without recognizing and respecting the power in concrete baby steps forward. Students are likewise ill-equipped to spend their energies designing concert halls, or full projects. It is precisely the accumulated experience of consulting and collaborating with architects, engineers of all kinds, and owners that allows for confidence when leaping into the gap between judgement and certainty. In honor of Knudsen's contributions to scientific exploration and education, we will dedicate ourselves to the betterment of our profession through real connections between academia and the consulting community.

THURSDAY MORNING, 25 OCTOBER 2012 COLONIAL BALLROOM, 10:30 A.M. TO 11:35 A.M.

Session 4aAAb

Architectural Acoustics, Noise, and ASA Committee on Standards: Acoustics and Health

David M. Sykes, Cochair
The Remington Group LP, 23 Buckingham St., Cambridge, MA 02138

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Institute, Troy, NY 12180

Invited Paper

10:30

4aAAb1. Hospital noise and staff performance. Gabriel Messingher, Erica Ryherd (Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu), and Jeremy Ackerman (Emergency Medicine, Emory University, Atlanta, GA)

Hospitals are often noisy and not conducive to staff performance. Indeed, many staff believe that noise negatively affects their professional performance, quality of work, and ability to concentrate and communicate. Research shows that increased stress and annoyance, increased rates of burnout, and reduced occupational health are a few of the possible effects of hospital noise on staff. However, only a few hospital studies have directly linked noise to job performance. Results show that noise and distractions can potentially

deteriorate mental efficiency and short-term memory and increase errors, but other studies have shown no significant effects. Alarm fatigue is also of concern, as staff may tune out, silence, or disable alarms because they are desensitized or exhausted by them. This paper will discuss what is currently known about hospital noise and staff performance and what questions remain. On-going studies relating the sound environment to staff performance in medical simulations will also be highlighted.

Contributed Papers

10:50

4aAAb2. Patient and staff perceptions of hospital noise. Nicola J. Shiers, Bridget M. Shield (Urban Engineering, London South Bank University, Borough Road, London SE1 7JQ, United Kingdom, shieldbm@lsbu.ac.uk), and Rosemary E. Glanville (Medical Architecture Research Unit, London South Bank University, London, United Kingdom)

A large scale survey of noise and acoustic conditions in a range of inpatient hospital wards has been undertaken in two major hospitals in the UK. The survey involved noise and acoustic surveys of occupied hospital wards, identification of noise sources and questionnaire surveys of nursing staff and patients. The surveys were carried out in a range of different ward types, including surgical and medical wards, and ward sizes. In total 25 patient bays were measured, varying in size from single rooms to large bays containing 12 beds. Questionnaire responses were received from 66 staff and 154 patients in the two hospitals. This paper will present the results of the questionnaire surveys relating to noise annoyance and disturbance among staff and patients. Factors which affect perceptions of noise will be examined including personal factors such as age, sex, and length of time working/staying in the hospital. The sources of noise which cause the most disturbance to staff and patients will also be discussed.

11:05

4aAAb3. A different perspective on the ongoing noise problem in U.S. hospitals: Lessons learned from existing acute care facilities and their patients' quiet-at-night scores. Gary Madaras (Making Hospitals Quiet, 4849 S. Austin Ave., Chicago, IL 60638, DoctorSonics@aol.com)

Acute care hospitals that care for Medicare patients now participate in the Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS) quality survey as part of The Hospital Value-Based Purchasing program implemented by the Centers for Medicare and Medicaid Services (CMS). One question on the 27-item survey asks inpatients to score "how often was the area around your room quiet at night" as 'always', 'usually', 'sometimes' or 'never'. Patients score the quietness question the lowest of all the quality metrics, responding only 58% of the time that the area around their room was always quiet at night (as compared to an average score of 72% for all other metrics). Results of the HCAHPS survey will affect market share and financial reimbursements from CMS. Hospitals are scrambling to reduce noise levels and increase HCAHPS scores. A study was conducted, asking leaders of hospitals to share their noise reduction stories. Leaders from 241 hospitals contributed their challenges, successes and lessons learned. This presentation will share the findings including an in-depth look at one of the participating hospitals. Further insight into the ongoing noise problem in hospitals will be gained via HCAHPS scores analysis and overnight noise audits recently conducted in existing hospitals.

11:20

4aAAb4. Designing quiet, healthy ductwork. Stephanie Ayers (Evonik Foams, Inc., Allentown, TX) and Michael Chusid (Chusid Associates, 18623 Ventura Blvd. #212, Tarzana, CA 91356, michael@chusid.com)

Acoustical duct liners promote a healthier interior environment by suppressing mechanical noise from heating, ventilating, and air conditioning (HVAC) systems. However the materials used to reduce or control noise may, themselves, have health implications. Fibrous acoustical insulation, for example, can release fibers into the air stream during installation or maintenance and when subjected to high velocity air or vibration. Recent studies have determined that glass fiber - the most prevalent duct liner material - should not be listed as a carcinogen. However, glass fiber is an acknowledged irritant. Moreover, long-term effects on sensitive populations - including children and individuals with compromised immune systems - have not been studied. Fibrous insulation can collect dust, thereby providing a site for mold and microbial growth. And dislodged particles can disturb sensitive electronics and clean room conditions. Some owners of facilities such as hospitals, schools, and laboratories have, therefore, prohibited use of fibrous acoustical liners in ductwork. This paper discusses the application of acoustical duct liners, and the performance and use of alternatives to glass fiber in situations where non-fibrous liners are required.

Session 4aABa

Animal Bioacoustics, Acoustical Oceanography, Structural Acoustics and Vibration, Underwater Acoustics, and ASA Committee on Standards: Underwater Noise from Pile Driving I

Mardi C. Hastings, Cochair
George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405

Martin Siderius, Cochair
ECE Dept., Portland State Univ., Portland, OR 97201

Chair's Introduction—8:50

Invited Papers

9:00

4aABa1. Experience measuring underwater sounds from piling activities. James A. Reyff (Illingworth & Rodkin, Inc., 505 Petaluma Blvd. South, Petaluma, CA 94952, jreyff@illingworthrodkin.com)

Extensive acoustic monitoring of pile driving activities along the U.S. West Coast has occurred in recent years as response to concerns regarding the effects to aquatic species. Impact pile driving activities have been found to produce high amplitude sounds that have injured fish and harassed marine mammals. As a result, requirements to reduce sounds, restrict the amount of pile driving and monitor effects to the environment have been required. The monitoring requirements vary for each project depending on the strength of the sound sources and potential presence of sensitive aquatic species. This presentation describes our experiences measuring acoustic signals from pile driving activities for various construction projects. Some results from testing sound attenuation devices are also presented. The challenges associated with monitoring these sounds are described, which include the complexities of measuring highly dynamic sounds in an environment with varying background levels. This presentation also describes the analysis methods used to describe pile driving sounds and how they are used to assess potential impacts to aquatic species. Methods for reporting results on a real-time or daily basis are also described.

9:20

4aABa2. Underwater radiated noise and impact assessment of marine piling operations during offshore windfarm construction. Paul A. Lepper (School of Electronic, Electrical and Systems Engineering, Loughborough University, Loughborough LE113TU, United Kingdom, p.a.lepper@lboro.ac.uk), Stephen P. Robinson, and Pete D. Theobald (National Physical Laboratory (NPL), Teddington, Middlesex, United Kingdom)

In UK waters numerous large scale offshore wind farm developments have been constructed typically using large hollow steel mono-pile foundations with pile diameters varying from a few meters to greater than 6 m diameter and lengths 60-80 m. Piles may be driven 20-30 m into the seabed in water depths from a few meters to greater than 30 m. Typically percussive piling construction operations are used with many thousands of individual strikes over periods of several hours resulting in repetitive high amplitude impulsive sound within the water column that has potential for impact on marine life. Data is presented for in-situ measurements made during installation a range of mono-pile diameters used on offshore windfarms. Full piling sequences were recorded at fixed ranges using fixed autonomous data loggers and sampled range dependent boat based measurements. Simultaneous recordings at multiple ranges varying from 10's meters to 10's km were made. Data is analyzed in terms of received levels, spectral and temporal components. Using range dependent propagation loss modeling equivalent mono-pole source levels are estimated. Level dependence on range, hammer energy, etc. are discussed. A Monte Carlo approach is used to obtain total cumulative exposure (SEL) risk for single foundation to whole windfarm construction scenarios.

9:40

4aABa3. On the Mach wave effect in impact pile driving, its observation, and its influence on transmission loss. Peter H. Dahl (Applied Physics Laboratory, Mechanical Engineering, University of Washington, 1013 NE 40th St, Seattle, WA 98105, dahl@apl.washington.edu) and Per G. Reinhall (Mechanical Engineering, University of Washington, Seattle, WA)

Pile driving in water produces extremely high sound pressure levels in the surrounding underwater environment of order 10 kPa at ranges of order 10 m from the pile that can result in deleterious effects on both fish and marine mammals. In Reinhall and Dahl [J. Acoust. Soc. Am. 130, 1209-1216, Sep. 2011] it is shown that the dominant underwater noise from impact driving is from the Mach wave associated with the radial expansion of the pile that propagates down the pile at speeds in excess of Mach 3 with respect to the underwater sound speed. In this talk we focus on observations of the Mach wave effect made with a 5.6 m-length vertical line array, at ranges 8-15 m in waters of depth ~12.5 m. The key observation is the dominant vertical arrival angle associated with the Mach wave, ~17 deg., but other observations include: its frequency dependence, the ratio of purely waterborne energy compared with that which emerges from the sediment, and results of a mode filtering operation which also points to the same dominant angle. Finally, these observations suggest a model for transmission loss which will also be discussed. [Research supported by the Washington State Department of Transportation.]

10:00–10:20 Break

10:20

4aABa4. Attenuation of pile driving noise using a double walled sound shield. Per G. Reinhall (Mechanical Engineering, University of Washington, MS 352600, Seattle, WA 98125, reinhall@uw.edu) and Peter H. Dahl (Applied Physics Laboratory, University of Washington, Seattle, WA)

Pile driving in water produces high sound levels in underwater environments. The associated pressures are known to produce deleterious effects on both fish and marine mammals. We present an evaluation of the effectiveness of surrounding the pile with a double walled sound shield to decrease impact pile driving noise. Four 32 m long, 76 cm diameter piles were driven 14 m into the sediment with a vibratory hammer. A double walled sound shield was then installed around the pile, and the pile was impact driven another 3 m while sound measurements were obtained. The last 0.3 m was driven with the sound shield removed, and data were collected for the untreated pile. The sound field obtained by finite element analysis is shown to agree well with measure data. The effectiveness of the sound shield is found to be limited by the fact that an upward moving Mach wave is produced in the sediment after the first reflection of the deformation wave against the bottom end of the pile. The sound reduction obtained through the use of the sound shield, as measured 10 meters away from the pile, is shown to be approximately 12dB dB re 1 μPa.

10:40

4aABa5. Transient analysis of sound radiated from a partially submerged cylindrical pile under impact. Shima Shahab and Mardi C. Hastings (George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

Underwater noise generated by impact pile driving has potentially harmful effects on aquatic animals and their environment. In effort to predict sound radiation from piling activities, a structural acoustics finite-difference, time-domain (FDTD) model has been developed for transient analysis of a partially submerged cylindrical pile. Three coupled partial differential equations govern vibration of the pile wall and six partial differential equations govern its boundary conditions. The space-time gridding underlying the numerical computations controls selection of an appropriate time step while the physical geometry of the pile imposes an upper limit on the frequency bandwidth of wall oscillations and radiated sound. This bandwidth is inversely proportional to diameter for a cylindrical steel pile. The higher the frequency content in the dynamic response, the smaller the time step required in a transient analysis. So as diameter of the pile decreases, smaller time steps are required to capture the total bandwidth observed in field data. Results of correlations between radiated sound predicted by the FDTD model and acoustic field data from piles of different diameter are presented. [Work supported by the Georgia Institute of Technology and Oregon Department of Transportation through a subcontract from Portland State University.]

11:00

4aABa6. Modeling underwater impact pile driving noise in shallow, inhomogeneous channels. Nathan D. Laws, Lisa Zurk, Scott Schecklman, and Martin Siderius (Electrical and Computer Engineering, Portland State University, Portland, OR 97210-3038, laws.nathan@gmail.com)

The broadband synthesis of a parabolic equation (PE) propagation model for shallow water acoustic propagation in inhomogeneous channels is presented to account for the noise produced by impact pile driving. The PE model utilizes sediment information obtained from boring measurements and detailed bathymetry to model range dependent propagation in the Columbia River between Portland, OR and Vancouver, WA. The impact pile driving source is modeled in two ways: first, as a reverberating impulse source that emits Mach-wave radiation [Reinhall, Dahl, J. Acoust. Soc. Am. 130, 1209 (2011)]; and second, with a structural acoustic finite-difference time-domain (FDTD) model [Shahab, Woolfe, and Hastings, J. Acoust. Soc. Am. 130, 2558 (2011)]. Model results using both source models are shown to be in good agreement with acoustic measurements of test pile operations in the Columbia River at multiple locations from 10 to 800 meters from the pile driving source. Implications for noise levels in river systems with varying bottom sediment characteristics are presented and discussed. [This research is supported with funding from the Oregon Department of Transportation.]

11:20

4aABa7. Underwater sound from pile driving and protected marine species issues. Amy R. Scholik-Schlomer (Office of Protected Resources, NOAA's National Marine Fisheries Service, 1315 East-West Hwy, SSMC3, Room 13605, Silver Spring, MD 20910, amy.scholik@noaa.gov) and Jason Gedamke (Office of Science and Technology, NOAA's National Marine Fisheries Service, Silver Spring, MD)

With current, wide-spread coastal construction projects and the predicted development of offshore wind energy, there are concerns regarding the potential impacts of underwater sound associated with pile driving activities on protected marine species. The National Marine Fisheries Service (NMFS) works to conserve, protect, and recover a variety of marine species, including marine mammals, marine and anadromous fishes, and sea turtles, protected under the Marine Mammal Protection Act (MMPA) and/or Endangered Species Act (ESA). In order to make management decisions for these protected species, we rely on scientific data to inform our policy. However, there are many challenges, including determining appropriate acoustic criteria and metrics for injury and behavioral harassment for impact and vibratory pile driving activities; understanding acoustic propagation in complex environments, especially shallow, coastal areas and throughout sediments; establishing appropriate protocols to mitigate and monitor impacts; and managing uncertainty for the broad number of species under our jurisdiction, who use and depend on sound (pressure and particle motion) in a variety of ways. Thus, we work collaboratively with other federal, state, and local government agencies, academia, nongovernmental agencies, and industry to best assess and manage risk from these activities.

11:40

4aABa8. Barotrauma effects on fishes in response to impulsive pile driving stimuli. Brandon M. Casper (Department of Biology, University of Maryland, College Park, MD 20742, bcasper@umd.edu), Michele B. Halvorsen, Thomas J. Carlson (Marine Sciences Laboratory, Pacific National Northwest Laboratory, Sequim, WA), and Arthur N. Popper (Department of Biology, University of Maryland, College Park, MD 20742)

We report on new results from controlled exposure studies of impulsive pile driving stimuli using the High Intensity Controlled Impedance Fluid Filled Wave Tube (HICI-FT). Following upon initial investigations focusing on injury thresholds and recovery from injuries in the Chinook salmon, experiments have been expanded to include lake sturgeon, Nile tilapia, hybrid striped bass, and hogchoker.

Several key questions concerning pile driving exposure in fishes have been explored utilizing species with different types of swim bladders as well as a species without a swim bladder. Injury thresholds were evaluated in all species, with recovery from injuries measured in the hybrid striped bass. Other pile driving variables measured with the hybrid striped bass include difference in response between fish less than or greater than 2g as well as the minimum number of pile strikes needed for injuries to appear. A study to evaluate potential damage to inner ear hair cells was also conducted on hybrid striped bass as well as the Nile tilapia. These studies will be utilized to better understand, and possibly predict, the potential effects of pile driving exposure in fishes.

THURSDAY MORNING, 25 OCTOBER 2012 LESTER YOUNG A, 8:55 A.M. TO 11:40 A.M.

Session 4aABb

Animal Bioacoustics: Terrestrial Passive Acoustic Monitoring I

David Delaney, Chair
U.S. Army CERL, Champaign, IL 61821

Chair's Introduction—8:55

Invited Papers

9:00

4aABb1. Acoustical monitoring of resource conditions in U.S. National Parks. Kurt M. Fristrup, Emma Lynch, and Damon Joyce (Natural Sounds and Night Skies Division, National Park Service, 1201 Oakridge Drive, Suite 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

Several laws and derived policy direct the National Park Service to conserve and restore acoustic resources unimpaired for the enjoyment of future generations. The Natural Sounds and Night Skies Division has collected acoustical and related meteorological data at more than 300 sites in over 60 park units spanning the coterminous U. S., with additional sites in Alaska, Hawaii, and American Samoa. Analyses of these data reveal that background sound levels in many park units approach or fall below the human threshold of hearing, and that noise intrusions are ubiquitous. An emergent challenge is to develop efficient tools to reprocess these data to document bioacoustical activity. Generic indices of wildlife activity would be useful for examining responses to climate change, other anthropogenic disturbance, and changes in park unit management. Documentation of individual species occupancy and calling density would inform identification of habitat characteristics and management of species of special concern.

9:20

4aABb2. Sensor arrays for automated acoustic monitoring of bird behavior and diversity. Charles Taylor (Ecology and Evolutionary Biology, UCLA, Los Angeles, CA 90064, taylor@biology.ucla.edu)

There is growing interest in how to automate analysis of acoustic monitoring of bird vocalizations – especially for monitoring bird behavior and biodiversity. I will review some of the main approaches to this problem and describe how this is being approached in our laboratory. We break the problem down to: event recognition; classification; localization; and analysis. These are not entirely independent. I will discuss some new approaches to these problems that seem to hold special promise.

9:40

4aABb3. Accurate localization over large areas with minimal arrays. Douglas L. Jones (Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, 1308 W. Main St., Urbana, IL 61801, dl-jones@uiuc.edu), Aaron L. Jones (Sonistic, LLC, Champaign, IL), and Rama Ratnam (Biology, University of Texas at San Antonio, San Antonio, TX)

Passive terrestrial acoustic monitoring often requires accurate localization of acoustic sources over large areas. At least five microphones are required for unambiguous 3D relative-time-delay-based localization, but within what range can reasonable accuracy practically be obtained? An efficient new method for estimating localization error for any given array geometry and location allows rapid exploration of the expected accuracy for any given array geometry and region. The accuracy is proportional to the standard deviation of the relative-time-delay error and an array-geometry-and-source-location-specific term. Box-like arrays show high accuracy within the box boundaries, but a quasi-linear "discrete helical" array also shows excellent planar localization performance, over large squarish regions of the extent of the long axis of the array to either side of that axis, independent of the array length. Assuming a 1 kHz bandwidth acoustic source with an approximate "Rayleigh" correlation error of 1 ms, the analysis shows the planar localization error to be less than 2 m within that region. Field tests with a 60-m five-element discrete helical array and several recorded bird and mammal calls closely conformed to the analytical estimates and experimentally achieved sub-2-m accuracy within 60 m of the array.

10:00–10:20 Break

10:20

4aABb4. An auditory approach to understanding the effects of noise on communication in natural environments. Robert Dooling, Sandra Blumenrath, Ed Smith, Ryan Simmons (Psychology, Univ of Maryland, Baltimore Ave, College Park, MD 20742, rdooling@umd.edu), and Kurt Fristrup (Natural Sounds Program, National Park Service, Fort Collins, CO)

Animals, like humans, frequently communicate using long-range acoustic signals in networks of several individuals. In socially and acoustically complex environments, however, communication is characterized by a variety of perceptual challenges that animals strive to overcome in order to interact successfully with conspecifics. Species differences in auditory sensitivity and the characteristics of the environment are major factors in predicting whether environmental noise limits communication between animals or interferes with detection of other biologically important sounds. Working with both birds and humans and using both synthetic and natural noises in both laboratory and field tests, we have developed a model for predicting the effects of particular masking noises on animal communication. Moreover, by comparing birds listening to bird vocalizations in noise with humans listening to speech in noise, we gain a novel intuitive feel for the challenges facing animals in noisy environments. This approach of considering communication from the standpoint of the receiver provides a better approach for understanding the effects of anthropogenic noises that exceed ambient levels. For instance, in determining risk to a particular species, effective communication distances derived from this model might be compared to other aspects of the species biology such as territory size.

10:40

4aABb5. A wireless acoustic sensor network for monitoring wildlife in remote locations. Matthew W. McKown (Ecology and Evolutionary Biology, Center for Ocean Health, UC Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, mwmckown@ucsc.edu), Martin Lukac (Nexleaf Analytics, Los Angeles, CA), Abraham Borker, Bernie Tershy, and Don Croll (Ecology and Evolutionary Biology, Center for Ocean Health, UC Santa Cruz, CA)

Seabirds are the most threatened marine group with nearly 28% of extant species considered at risk of extinction. Managers and researchers face considerable financial and logistical challenges when designing programs to monitor the status of any of the 97 species listed as critically endangered, endangered, or vulnerable by the IUCN. These challenges are exacerbated by the fact that these birds breed in isolated/inaccessible locations, many have cryptic nest sites, and most return to colonies only at night. Acoustic sensors are an effective tool for monitoring the presence, distribution, and relative abundance of rare and elusive seabirds. We have developed new, cellphone-based wireless acoustic sensors that 1) are comparable to state-of-the-art sensors, 2) are affordable (~US$500.00 per hectare), 3) can sample continuously over months, 4) can telemeter data from remote locations via a cellular, microwave, or satellite link, and 5) can be reprogrammed remotely. To date we have deployed our wireless acoustic sensor networks to monitor seabirds of conservation concern including - Ashy Storm-petrel, *Oceanodroma homochroa*, on Southeast Farallon Island (CA), Tristram's Storm-petrel, *O. tristrami*, on Tern Island (French Frigate Shoals), as well as Newell's Shearwater, *Puffinus newelli*, and Hawaiian Petrel, *Pterodroma sandwichensis*, at the Upper Limahuli Preserve (Kaua'i, HI).

11:00

4aABb6. A template-based automatic bird phrase classification in noisy environments using limited training data. Kantapon Kaewtip, Lance Williams, Lee N. Tan (Electrical Engineering, UCLA, Los Angeles, CA), George Kossan (Ecology and Evolutionary Biology, UCLA, Los Angeles, CA), Abeer Alwan (Electrical Engineering, UCLA, Los Angeles, CA), and Charles Taylor (Ecology and Evolutionary Biology, UCLA, Los Angeles, CA 90064, taylor@biology.ucla.edu)

Bird Songs typically comprise a sequence of smaller units, termed phrases, separated from one another by longer pauses; songs are thought to assist in mate attraction and territory defense. Studies of bird song would often be helped by automated phrase classification. Past classification studies usually employed techniques from speech recognition, such as MFCC feature extraction and HMMs. Problems with these methods include degradation from background noise, and often require a large amount of training data. We present a novel approach to robust bird phrase classification using template-based techniques. One (or more) template is assigned to each phrase with its specific information, such as prominent time-frequency components. In our trials with 1022 phrases from Cassin's Vireo (Vireo cassinii) that had been hand-identified into 32 distinct classes, far fewer few examples per class were required for training in some cases only 1 to 4 examples for 84.95%-90.27% accuracy. The choice of distance metrics was crucial for such systems. We found that weighted 2D convolution is a robust distance metric for our task.. We also studied phrase patterns using Multi-Dimensional Scaling, a discriminative feature for phrase patterns that are very similar

11:20

4aABb7. Separating anthropogenic from natural sound in a park setting. John Gillette, Jeremy Kemball, and Paul Schomer (Schomer and Associates, 2117 Robert Drive, Champaign, IL 61821, gillett1@illinois.edu)

This paper is a continuation of a study with the National Park Service to detect and separate natural and anthropogenic sound in a park setting. Last year, an algorithm was written to detect anthropogenic tones, because virtually all anthropogenic sound contains tones less than one 1KHz. By comparing each frequency to the surrounding third octave band, the algorithm detects almost all anthropogenic sounds. However, this method does not work for jet aircraft, because of their broad band sound when flying at altitude, not tones. This year, an algorithm was developed to detect jet aircraft, by comparing natural and anthropogenic sound over time as opposed to over frequency. The algorithm finds average equivalent sound level (LEQ) over a day and then removes anomalous peaks from the original sound recording. The program then subtracts the LEQ from the entire file, and the remaining sound is marked as probable jet aircraft sound. If the sound is present for a long enough duration, it is recorded as a jet sound.

Session 4aBA

Biomedical Acoustics and Physical Acoustics: Cavitation in Biomedical and Physical Acoustics

Xinmai Yang, Chair
Mechanical Engineering, Univ. of Kansas, Lawrence, KS 66045

Contributed Papers

8:00

4aBA1. Effects of encapsulation damping on frequency dependent subharmonic threshold for contrast microbubbles. Amit Katiyar (Mechanical Engineering, University of Delaware, Newark, DE) and Kausik Sarkar (Mechanical and Aerospace Engineering, George Washington University, 801 22nd Street NW, Washington, DC 20052, sarkar@gwu.edu)

The frequency of minimum threshold for subharmonic generation from contrast microbubbles is investigated here. Increased damping—either due to the small radius or the encapsulation—is shown to shift the minimum threshold away from twice the resonance frequency. Free bubbles as well as four different models of the contrast agent encapsulation are investigated varying the surface dilatational viscosity. Encapsulation properties are determined using measured attenuation data for a commercial contrast agent. For sufficiently small damping, models predict two minima for the threshold curve—one at twice the resonance frequency being lower than the other at resonance frequency—in accord with the classical analytical result. However, increased damping damps the bubble response more at twice the resonance than at resonance, leading to a flattening of the threshold curve and a gradual shift of the absolute minimum from twice the resonance frequency towards the resonance frequency. The deviation from the classical result stems from the fact that the perturbation analysis employed to obtain it assumes small damping, not always applicable for contrast microbubbles (Supported by NSF CBET-0651912, CBET-1033256, DMR-1005283).

8:15

4aBA2. Pulse duration dependence of cavitation emissions and loss of echogenicity from ultrasound contrast agents insonified by Doppler pulses. Kirthi Radhakrishnan (Biomedical Engineering, University of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267, radhakki@mail.uc.edu), Kevin J. Haworth (Internal Medicine, University of Cincinnati, Cincinnati, OH), Jonathan A. Kopechek (Mechanical Engineering, Boston University, Boston, MA), Bin Huang (Division of Biostatistics and Epidemiology, Children's Hospital Medical Center, Cincinnati, OH), Shaoling Huang, David D. McPherson (Internal Medicine, University of Texas Health Science Center, Houston, TX), and Christy K. Holland (Internal Medicine, University of Cincinnati, Cincinnati, OH)

Careful determination of stable and inertial cavitation thresholds of UCAs exposed to pulsed ultrasound is required for their safe use in diagnostic and therapeutic applications. Echogenic liposomes and Definity® were diluted in porcine plasma and pumped through a physiological flow phantom. UCAs were insonified with pulsed Doppler ultrasound at three pulse durations (3.33 μs, 5.83 μs and 8.33 μs) over a range of peak rarefactional pressure amplitudes (0.06-1.9 MPa). A 10-MHz focused passive cavitation detector (PCD) was used to record cavitation emissions. PCD signals and B-mode images of UCAs and degassed water were acquired during insonation. Thresholds of stable and inertial cavitation, and loss of echogenicity were determined by piecewise linear fits of the cavitation powers and mean gray scale values, respectively. The stable cavitation thresholds were found to be lower than the inertial cavitation thresholds at each pulse duration setting. The thresholds of loss of echogenicity and stable and inertial cavitation were found to be dependent on pulse duration. The relationship between loss of echogenicity and cavitation emissions will be discussed in the context of using onscreen echogenicity to indirectly monitor cavitation during ultrasound-mediated therapy with UCAs. [Supported by NIH R01 HL059586.]

8:30

4aBA3. Echogenicity and release characteristics of folate-conjugated echogenic liposomes for cytosolic delivery of cancer drugs. Shirshendu Paul (Mechanical Engineering, University of Delaware, Newark, DE), Rahul Nahire, Sanku Mallik (Pharmaceutical Sciences, North Dakota State University, Fargo, ND), and Kausik Sarkar (Mechanical and Aerospace Engineering, George Washington University, 801 22nd Street NW, Washington, DC 20052, sarkar@gwu.edu)

Echogenic liposomes (ELIPs) are specially prepared liposomes that encapsulate both aqueous and gaseous phases. The presence of gas makes them echogenic. Since, ELIPs retain all the favorable properties of normal liposomes they can be used for simultaneous ultrasonic imaging and drug delivery applications. These liposomes are polymerized on the external leaflet using a disulphide linker. Disulphide bonds are reversibly broken in presence of thiol above a critical concentration. Therefore, the liposomes are stable in the plasma (thiol concentration 10 μM) but release its content inside the cell (thiol concentration 10 mM). The liposome also expresses folate group on its surface which allows its entry into the cancer cells. The release can be controlled by diagnostic frequency ultrasound. Therefore, these ELIPs hold promises for ultrasound image-guided cytosolic delivery for cancer drugs. We will report on their acoustic properties and ultrasound-mediated release characteristics. Their implications on design and development of these novel contrast agents will be discussed. [Supported by NSF CBET-0651912, CBET-1033256, DMR-1005283.]

8:45

4aBA4. High-frequency harmonic imaging with coded excitation: Implications for the assessment of coronary atherosclerosis. Himanshu Shekhar and Marvin M. Doyley (Department of Electrical and Computer Engineering, University of Rochester, Rochester, NY 14611, himanshushekhar@rochester.edu)

The adventitial *vasa vasorum* grows abnormally in life-threatening atherosclerotic plaques. Harmonic intravascular ultrasound (H-IVUS) could help assess the *vasa vasorum* by nonlinear imaging of microbubble contrast agents. However, the harmonics generated in tissue at high acoustic pressures compromise the specificity of H-IVUS - a trait that has hampered its clinical use. Therefore, H-IVUS should be conducted at low pressure amplitudes; but the resulting decrease in signal-to-noise ratio (SNR) could limit the sensitive detection of the *vasa vasorum*. In this study, we investigated the feasibility of improving the SNR of H-IVUS imaging with chirp-coded excitation. Numerical simulations and experiments were conducted to assess the harmonic response of the commercial contrast agent Targestar-p^{TM}, to sine-burst and chirp-coded excitation (center frequencies 10 and 13 MHz, peak-pressures 100 to 300 kPa). We employed 1) a single-element transducer pair, and 2) a dual-peak frequency transducer for our studies. Our experimental results demonstrated that exciting the agent with chirp-coded pulses can improve the harmonic SNR by 7 to 14 dB. Further, the axial resolution obtained with chirp-coded excitation was within 10% of that expected for sine-burst excitation. Therefore, we envisage that chirp-coded excitation may be a viable strategy to visualize the *vasa vasorum* with H-IVUS imaging.

9:00

4aBA5. Effect of inter-element apodization on passive cavitation images. Kevin J. Haworth (Internal Medicine, University of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209, kevin.haworth@uc.edu), T. D. Mast, Kirthi Radhakrishnan (Biomedical Engineering Program, University of Cincinnati, Cincinnati, OH), and Christy K. Holland (Internal Medicine, University of Cincinnati, Cincinnati, OH)

Acoustic cavitation has been correlated with a variety of ultrasound-mediated bioeffects. Recently developed passive cavitation imaging methods provide spatially resolved maps of cavitation activity with good azimuthal resolution but poor axial resolution. Here, inter-element apodization is investigated as a means of improving image quality. Cavitation was induced from echogenic liposomes in a flow phantom exposed to 6 MHz Doppler ultrasound (Philips HDI-5000). The resulting acoustic emissions were passively recorded on 64 elements of a linear array (L8-3 transducer, Zonare z.one ultra scanner). Amplitude scaling of each waveform by its root-mean-square value improved axial resolution at the expense of creating an 'X-shaped' artifact. Cosine amplitude apodization of the received waveforms across the array and centered about the azimuthal location of the beamformed image pixel was found to reduce grating lobe artifacts. Numerical time reversal of the received waveforms, using the Fresnel approximation for the acoustic field of each array element, resulted in an effective apodization due to element directivity and also reducing grating lobe artifacts. Applying apodization may be an effective means of increasing passive image quality for certain cavitation distributions, which will be discussed. [Supported in part by NIH grants F32HL104916, R01HL074002, R21EB008483, R01HL059586, and R01NS047603.]

9:15

4aBA6. Acoustic emissions associated with ultrasound-induced rupture of *ex vivo* blood vessels. Cameron L. Hoerig (Electrical Engineering Program, University of Cincinnati, Cincinnati, OH), Joseph C. Serrone (Department of Neurosurgery, University of Cincinnati, Cincinnati, OH), Mark T. Burgess (Biomedical Engineering Program, University of Cincinnati, Cincinnati, OH), Mario Zuccarello (Department of Neurosurgery, University of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Engineering Program, University of Cincinnati, 3938 Cardiovascular Research Center, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Occlusion of blood vessels using high-intensity focused ultrasound (HIFU) is a potential treatment for arteriovenous malformations and other neurovascular disorders. However, HIFU-induced vessel occlusion can cause vessel rupture resulting in hemorrhage. Possible rupture mechanisms include mechanical effects of acoustic cavitation and hyperthermia of the vessel wall. To investigate the mechanism of vessel rupture and assess the possibility of rupture prediction from acoustic emissions, HIFU exposures were performed on 18 *ex vivo* porcine femoral arteries with simultaneous passive cavitation detection. Vessels were insonified by a 3.3 MHz focused source with spatial-peak, temporal-peak focal intensity 1728-2791 W/cm^2 and a 50% duty cycle for durations up to 5 minutes. Time-dependent acoustic emissions were recorded by an unfocused passive cavitation detector and quantified within low-frequency (10-30 kHz), broadband (0.3-1.1 MHz), and subharmonic (1.65 MHz) bands. Vessel rupture was detected by inline metering of saline flow, recorded throughout each treatment. Rupture prediction tests, using receiver operating characteristic curve analysis, found subharmonic emissions to be most predictive. These results suggest that acoustic cavitation plays an important role in HIFU-induced vessel rupture. In HIFU treatments for vessel occlusion, passive monitoring of acoustic emissions may be useful in avoiding hemorrhage.

9:30

4aBA7. Cavitation mechanisms in ultrasound-enhanced permeability of *ex vivo* porcine skin. Kyle T. Rich (Biomedical Engineering Program, University of Cincinnati, Cincinnati, OH), Cameron L. Hoerig (Electrical Engineering Program, University of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Engineering Program, University of Cincinnati, 3938 Cardiovascular Research Center, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, doug.mast@uc.edu)

Ultrasound-induced cavitation is known to enhance transdermal transport of drugs for local and systemic delivery. However, the specific cavitation mechanisms responsible are not well understood, and the physical location of permeability-enhancing cavitation is also unknown. The experiments reported here investigated the role of stable and inertial cavitation, both within the skin and at the dorsal skin surface, in ultrasound enhancement of skin permeability. Full-thickness porcine skin was hydrated with either air-saturated phosphate buffered saline (PBS) or vacuum-degassed PBS to localize cavitation activity within or outside the skin, respectively. Skin samples were sonicated for 30 minutes over a range of frequencies (0.41 and 2.0 MHz) and peak rarefactional pressure amplitudes (0-750 kPa) with a 20% duty cycle (1 s on, 4 s off). Cavitation activity was monitored using a 1.0 MHz unfocused, wideband passive cavitation detector (PCD). Changes in skin permeability were quantified by measuring the electrical resistance of skin every 10 seconds during insonation. Subharmonic acoustic emissions revealed a strong correlation with decreasing electrical resistance of skin when cavitation was isolated within the tissue, suggesting that stable cavitation within the skin plays a primary role in ultrasound-enhanced permeability over the frequencies investigated.

9:45

4aBA8. Laser-induced-cavitation enhanced ultrasound thrombolysis. Huizhong Cui and Xinmai Yang (Mechanical Engineering, University of Kansas, 5109 Learned Hall, Lawrence, KS 66045, xmyang@ku.edu)

Application of ultrasound (US) is considered as an effective way to dissolve thrombus. Cavitation has been demonstrated to be significant to enhance thrombolytic efficacy. In this study, to improve the efficacy of this thrombolytic therapy, 764-nm laser light was used to induce cavitation in the US thrombolysis. Porcine clots were cut into small pieces and inserted into small tubes, then placed in the focal zone of a 1-MHz high-intensity focused ultrasound (HIFU) transducer in a water tank. At the same time, a 10-Hz laser system, which is confocal with the HIFU transducer, was used to illuminate on the focal area of the model during thrombolysis. After thrombolysis, the debris of clots was weighed to calculate the weight loss. Both US thrombolysis with and without laser illumination were performed in the experiment. Different combinations of peak-to-peak ultrasound pressure amplitude, duty cycle and duration were used. It is shown that the clot mass loss increased significantly when the laser illumination presented during the US thrombolysis process. The preliminary experimental results indicated that laser induced cavitation may play an important role in the enhancement of US thrombolysis.

10:00–10:30 Break

10:30

4aBA9. Ethanol injection induced cavitation and heating in tissue exposed to high intensity focused ultrasound. Chong Chen (Department of Biomedical Engineering, Tulane University, New Orleans, LA), Yunbo Liu, Subha Maruvada, Matthew Myers (Center for Devices and Radiological Health, U.S. Food and Drug Administration, Silver Spring, MD), and Damir Khismatullin (Department of Biomedical Engineering, Tulane University, 500 Lindy Boggs Center, New Orleans, LA 70118, damir@tulane.edu)

High Intensity Focused Ultrasound (HIFU) can ablate tumors located deep in the body through highly localized energy deposition and tissue heating at the target location. The volume of a HIFU-induced thermal lesion can be increased in the presence of cavitation. This study explores the effect of ethanol injection on cavitation and heating in tissue-mimicking phantoms and bovine liver tissues exposed to HIFU. The HIFU transducer (0.825 MHz) operated at seven acoustic power levels ranging from 1.3 W to 26.8 W. The cavitation events were quantified by B-mode ultrasound imaging, needle hydrophone measurements, and passive cavitation detection (PCD). Temperature in or near the focal zone was measured by thermocouples embedded in the samples. The onset of inertial cavitation in ethanol-treated phantoms and bovine liver tissues occurred at a lower power level than in the untreated samples (control). The cavitation occurrence in turn resulted in a sudden rise of temperature in ethanol-treated samples at a lower acoustic power than that in control. The results of this work indicate that the use of percutaneous ethanol injection prior to HIFU exposure may improve the HIFU therapeutic efficiency.

10:45

4aBA10. Scattering by bubbles at frequencies well below resonance.
R. L. Culver, Robert W. Smith, and Dale I. McElhone (ARL, Penn State University, PO Box 30, State College, PA 16804, rlc5@psu.edu)

We are interested in acoustic scattering by bubble clouds in water at frequencies and densities such that the acoustic wavelength is large relative to the average distance between bubbles and large relative to that corresponding to the bubble resonance frequency. At high frequency and moderate bubble density, bubble scattered intensity is proportional to N (the number density of the bubbles, m^4), which corresponds to incoherent scattering. Effective medium theory has been shown to predict predominantly incoherent scattering at high frequencies, but coherent scattering (scattered intensity proportional to N^2) at lower frequencies. An incoherent scattering assumption at low frequencies can substantially under predict the intensity of the scattered signal. Coherent (low frequency) scattering from bubble assemblages has also been explained in terms of collective shape, but this approach does not provide a means of predicting the temporal extent of the scattered signal in low frequency regimes. The literature apparently does not provide precise guidance as to when and how bubble scattering transitions from incoherent to coherent scattering in response to increasing wavelength, and the relationship between the acoustic wavelength and average bubble separation. Modeling and a tank experiment are underway that we hope will provide some answers to this question.

11:00

4aBA11. Low-frequency measurement of encapsulated bubble compressibility.
Scott J. Schoen, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Applied Research Laboratories, The University of Texas at Austin, 204 E. Dean Keeton Street, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Interest in measuring underground water flow has motivated synthesis of encapsulated microbubbles for use as contrast agents. The large acoustic attenuation in earth prohibits use of the high frequencies required to exploit resonant scattering. Instead, contrast enhancement must rely on the reduction of acoustic impedance due to higher compressibility of the microbubbles. Bubble compressibility is measured at the kilohertz frequencies of interest using a resonance tube filled with water and observing the change in tube resonance frequency due to the presence of bubbles for different void fractions [Wilson and Dunton, J. Acoust. Soc. Am. **125**, 1951 (2009)]. Buoyancy makes it difficult to maintain a uniform distribution of bubbles throughout the tube in order to relate sound speed to resonance frequency. Therefore, the bubbles were restrained with acoustically transparent barriers to form discrete layers within the water column. A model was developed to investigate the effect on the tube resonance frequency due to different spatial distributions of the bubble layers, and the predictions were compared with measurements. Good agreement with the known compressibility of air was obtained experimentally with only three or four layers. [Work supported by Advanced Energy Consortium.]

11:15

4aBA12. Measurements of resonance frequencies and damping of large encapsulated bubbles in a closed, water-filled tank.
Kevin M. Lee, Andrew R. McNeese, Laura M. Tseng, Mark S. Wochner (Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758, klee@arlut.utexas.edu), and Preston S. Wilson (Mechnical Engineering Department and Applied Research Laboratories, The University of Texas at Austin, Austin, TX)

The ultimate goal of this work is to accurately predict the resonance frequencies of large (on the order of 10 cm radius) tethered encapsulated bubbles used in an underwater noise abatement system, and also to investigate ways to enhance the system's efficacy over the use of air-filled bubbles alone. Toward that end, a closed water-filled tank was developed for the purpose of measuring the resonance frequency and damping of single large tethered encapsulated bubbles. The tank was designed to be operated in the long wavelength limit for frequencies below the lowest tank resonance, which was chosen to be 500 Hz, using the method described be Leighton, et al. [J. Acoust. Soc. Am. **112**, 1366–1376 (2002)]. Individual bubble resonance frequencies and *Q*-factors were measured for encapsulated bubbles of various sizes. The effects of the encapsulating material and wall thickness were investigated, along with the effects of alternative fill gases and internal heat transfer materials. Experimental results are compared with an existing predictive model [J. Acoust. Soc. Am. **97**, 1510–1521 (1995)] of bubble resonance and damping. [Work supported by Shell Global Solutions.]

11:30

4aBA13. Application of inversion techniques for bubble size spectra from attenuation measurements in lab-generated bubble distributions.
Dale I. McElhone, Robert W. Smith, and R. Lee Culver (Appl. Res. Lab., Penn State Univ., State College, PA 16804, dalemcsquared@gmail.com)

The size distribution of a bubble population can be estimated from measurements of the frequency-dependent attenuation through the bubble cloud. These attenuation values are the inputs to an inversion method that makes use of a resonant bubble approximation wherein it is assumed that only resonant bubbles contribute to the attenuation at a given frequency, *e.g*., Caruthers *et al*., [JASA (1999)]. H. Czerski has shown that power law bubble distributions proportional to $(\text{radius})^x$, where $x \leq -2$, have few enough large bubbles for resonant bubble inversion methods to yield accurate results. In this paper, the Caruthers and Czerski inversion methods are both verified with synthetic data and applied to acoustic measurements in a fresh water tank using lab-generated bubble distributions. Work sponsored by the Office of Naval Research, Code 321US.

Session 4aEA

Engineering Acoustics: Layered Media

Andrew J. Hull, Chair
Naval Undersea Warfare Center, Newport, RI 02841

Invited Papers

8:20

4aEA1. Acoustic radiation from a point excited multi-layered finite plate. Sabih I. Hayek (Engineering Science and Mechanics, Penn State University, State College, PA 16802, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (NAVSEA, Division Newport, Newport, RI)

The acoustic radiation from a finite rectangular composite plate is evaluated using eigenfunctions obtained through the use of three-dimensional equations of elasticity. The composite plate is made of perfectly bonded finite plates of identical lateral dimensions and of different thicknesses. The plate is free of shear stresses and is pinned on the in-plane displacements on all its boundaries and is baffled by an infinite rigid plane. The multi-layered plate is in contact with a different fluid medium on each of its two surfaces. The solution for the vibration response due to normal and shear surface forces is found in terms of the composite plate eigenfunctions that include heavy acoustic loading. The displacement vector field throughout the thickness of the plate is computed as well as the resultant near- and far-field radiated acoustic pressures for various ratios of thickness to plate dimensions over a broad frequency range. Initial results focus on a bilaminar plate. [Work supported by the ASEE Summer Faculty Research Program.]

8:40

4aEA2. Investigating the fidelity of a pseudo-analytical solution of a rib-stiffened, layered plate structure subjected to high frequency acoustic loading. Kirubel Teferra and Jeffrey Cipolla (Applied Science Division, Weidlinger Associates, 375 Hudson St., New York, NY 10014, kirubel.teferra@wai.com)

There is a need for a fast and reliable tool to assist in the analysis, design, and optimization of submarine and UUV coatings due to high frequency incident acoustic pressure loading. An existing pseudo-analytical, frequency domain solution for wave propagation in coated, ribbed, three-dimensional elastic layered plates excited by acoustic plane waves provides fast solutions for high frequency excitations. Weidlinger Associates, Inc. (WAI) is developing an analysis software tool which integrates this solution methodology while adding some technical improvements to the formulation. The solution methodology, which is found to be numerically unstable under certain conditions, contains a fundamental ansatz regarding the set of excited wave forms expressed through a particular wave number expansion in the direction of periodicity. Evidence is presented to show that the numerical instability is due to the specific choice of the wave number basis used in the solution. In order to provide a remedy while retaining the positive aspects of the solution methodology, WAI is implementing a pre-processing step to determine the optimal wave number basis: the set of admissible propagating (and attenuating) waves are predetermined via an eigenvalue analysis and then substituted into the wave number basis in computing the pseudo-analytical solution.

9:00

4aEA3. Elasto-acoustic response of a rib-stiffened multi-layer Hull system. Irena Lucifredi (SOFAR Acoustics, 44 Garfield Ave. #2, Woburn, MA 01801, euler001@yahoo.com), Raymond J. Nagem (Boston University, Boston, MA), and Federico Lucifredi (SOFAR Acoustics, Woburn, MA)

The analysis of hull vibrations has been a long-standing topic of interest in the US Navy for both surface and underwater vehicles. Understanding of the physics controlling acoustic scattering and radiation from coated, fluid-loaded structures is important as it can provide the required knowledge of the self-noise modeling of hull arrays and of the acoustic target strength of the submersibles. Currently, models are typically limited to low frequency regime of operation, not being able to consider a broad mid-high frequency range, commonly rich in physical phenomena that characterize sound fields in underwater vehicle environments. The goal of this effort is to provide a robust, innovative, and computationally efficient tool for analytical modeling of a fluid-loaded acoustic coating affixed to a rib-stiffened backing plate, capable of representing high frequency acoustic environments not suitable for conventional finite element approaches.The approach taken in this effort is based on the A.J. Hull's derivation of the elastic response of a layered sonar system on a rib-stiffened plate, and it is centered on the reformulation of the layered system response problem using displacement and stress variables.The new approach produces a significant improvement in the stability, efficiency, and accuracy of the computational method.

9:20

4aEA4. Vibro-acoustic response of an infinite, rib-stiffened, thick-plate assembly using finite-element analysis. Marcel C. Remillieux (Mechanical Engineering, Virginia Tech, 149 Durham Hall, Blacksburg, VA 24061, mremilli@vt.edu)

The vibration of and sound radiation from an infinite, fluid-loaded, thick-plate assembly stiffened periodically with ribs are investigated numerically using finite element (FE) analysis. The analysis is conducted in two-dimensions using plane-strain deformation to model the dynamics of the structure. Advantage is taken of the periodicity of the system to deal with the infinite dimensions of the model through the use of periodic boundary conditions. Firstly, numerical simulations are used to validate the analytical solutions derived

recently for this particular problem by Hull and Welch [Elastic response of an acoustic coating on a rib-stiffened plate, Journal of Sound and Vibration 329 (2010) 4192-4211]. Numerical and analytical solutions are in excellent agreement, provided that the number of modes in the analytical model is chosen correctly. Through this validation effort it is also demonstrated that the analytical model is sensitive to the number of modes used to formulate the solution, which may result in some instabilities related to mode count. Subsequently, the numerical model is used to study the effect of repeated and equally spaced void inclusions on the vibro-acoustic response of the system.

Contributed Papers

9:40

4aEA5. Radiation loading on multi-panel plates. Chiruvai P. Vendhan, Poosarla V. Suresh, and Subrata K. Bhattacharyya (Ocean Engineering Department, Indian Institute of Technology Madras, Chennai, Tamilnadu 600036, India, vendhan@iitm.ac.in)

Fluid-structure interaction problems involving the harmonic vibration of plates may be analyzed by employing an assumed modes approach. The associated hydrodynamic problem may be solved employing boundary element or finite element (FE) methods. An infinitely long multi-panel plate, having uniform spans and vibrating in contact with a fluid is considered here. A typical single span panel of the multi-panel system is set in a rigid baffle and the semi-infinite fluid domain over it is truncated. A FE model for the Helmholtz equation is employed over this domain, and suitable dampers are used on the truncation boundary to impose the radiation boundary condition. The FE solution is used to set up an eigenfunction expansion of the acoustic field outside the FE domain. Such an approach has originally been developed for exterior acoustic problems [C.P. Vendhan and C. Prabavathi, J. Vib. and Acoust., 118, 1996, 575-582]. The pressure field on the single panel and the infinite baffle is used to obtain the modal radiation loading in the form of added mass and radiation damping matrices of the multi-panel system, employing reciprocity and linear superposition. The method has been validated for an infinite plate example and illustrated using two and three panel systems.

9:55–10:15 Break

10:15

4aEA6. Free-wave propagation relationships of second-order and fourth-order periodic systems. Andrew J. Hull (Naval Undersea Warfare Center, 1176 Howell St, Newport, RI 02841, andrew.hull@navy.mil)

This talk develops an analytical expression for the determinant of two diagonally-indexed, full matrices when they are zero. These matrices originate from second- and fourth-order periodic system theory. The partial differential equations of these systems are solved using a series solution and are converted into closed-form analytical expressions. The denominators of these expressions are zero when free-wave propagation is present, and these denominators are equated to the determinants of the system matrices derived from a second analytical method. This process develops a relationship between frequency and wavenumber that is explicit for free-wave propagation in these systems. Two examples are included to illustrate this relationship.

10:30

4aEA7. Damping of flexural vibrations in glass fiber composite plates and honeycomb sandwich panels containing indentations of power-law profile. Elizabeth P. Bowyer, Peter Nash, and Victor V. Krylov (Aeronautical and Automotive Engineering, Loughborough University, Loughborough, Leicetershire LE11 3TU, United Kingdom, V.V.Krylov@lboro.ac.uk)

In this paper, the results of the experimental investigation into the addition of indentations of power-law profile into composite plates and panels and their subsequent inclusion into composite honeycomb sandwich panels are reported. The composite plates in question are sheets of composite with visible indentations of power-law profile. A panel is a sheet of composite with the indentations encased within the sample. This makes a panel similar in surface texture to an un-machined composite sheet (reference plate) or conventional honeycomb sandwich panel. In the case of quadratic or higher-order profiles, the above-mentioned indentations act as two-dimensional acoustic black holes for flexural waves that can absorb a large proportion of the incident wave energy. For all the composite samples tested in this investigation, the addition of two-dimensional acoustic black holes resulted in further increase in damping of resonant vibrations, in addition to the already substantial inherent damping due to large values of the loss factor for composites. Due to large values of the loss factor for composite materials used, no increase in damping was seen with the addition of a small amount of absorbing material to the indentations, as expected.

10:45

4aEA8. Sound radiation of rectangular plates containing tapered indentations of power-law profile. Elizabeth P. Bowyer and Victor V. Krylov (Aeronautical and Automotive Engineering, Loughborough University, Loughborough, Leicestershire LE11 3TU, United Kingdom, V.V.Krylov@lboro.ac.uk)

In this paper, the results of the experimental investigations into the sound of rectangular plates containing tapered indentations of power-law profile are reported. Such tapered indentations materialise two-dimensional acoustic black holes for flexural waves that result in absorption of a large proportion of the incident wave energy. A multi-indentation plate was compared to a plain reference plate of the same dimensions, and the radiated sound power was determined (ISO 3744). It was demonstrated that not only do such multiple indentations provide substantial reduction in the damping of flexural vibrations within the plates, but also cause a substantial reduction in the radiated sound power. As the amplitudes of the flexural vibrations of a plate are directly linked to the amplitude of radiated sound from the same plate, this paper also considers the effect of distribution of the amplitude of the plate's response on the amplitudes of the radiated sound. This investigation concludes that, despite an increase in the amplitude of the displacement at the indentation tip, the overall reduction in the constant thickness of the plate is large enough to result in substantial reductions in the overall vibration response and in the resulting sound radiation of plates containing indentations of power-law profile.

11:00

4aEA9. Damping of flexural vibrations in plates containing ensembles of tapered indentations of power-law profile. Elizabeth P. Bowyer, Daniel O'Boy, and Victor V. Krylov (Aeronautical and Automotive Engineering, Loughborough University, Loughborough, Leicestershire LE11 3TU, United Kingdom, V.V.Krylov@lboro.ac.uk)

In this work, we report experimental results on damping flexural vibrations in rectangular plates containing tapered indentations (pits) of power-law profile, the centres of which are covered by a small amount of absorbing material. In the case of quadratic or higher-order profiles, such indentations materialise two-dimensional acoustic 'black holes' for flexural waves. Initially, the effects of single pits have been investigated. It has been found that, in order to increase the damping efficiency of power-law profiled indentations, their absorption cross-sections should be enlarged by drilling a central hole of sufficiently large size (14 mm), while keeping the edges sharp. Such pits, being in fact curved power-law wedges, result in substantially increased damping. The next and the major part of this investigation involved using multiple indentations in the same rectangular plates to increase damping. Plates with combinations from two to six equal indentations have been investigated. The results show that, when multiple indentations are used, the associated damping increases substantially with the increase of a number of indentations. For the plate with 6 indentations, the resulting damping becomes comparable if not greater than that achieved by a wedge of power-law profile.

Session 4aMUa

Musical Acoustics and Speech Communication: The Acoustics of Rhythm

James P. Cottingham, Chair
Physics, Coe College, Cedar Rapids, IA 52402

Chair's Introduction—8:25

Invited Papers

8:30

4aMUa1. The nature and perception of human musical rhythms. Holger Hennig (Jefferson Lab, Dept. of Physics, Harvard University, Cambridge, MA 02138, holgerh@physics.harvard.edu), Ragnar Fleischmann (Nonlinear Dynamics, Max Planck Instittute for Dynamics and Self-Organization, Goettingen, Nds, Germany), Anneke Fredebohm (Institute of Psychology, University of Goettingen, Goettingen, Nds, Germany), York Hagmayer (King's College, Goettingen, Nds, Germany), Jan Nagler, Annette Witt (Nonlinear Dynamics, Max Planck Instittute for Dynamics and Self-Organization, Goettingen, Nds, Germany), Fabian J. Theis (Institute for Bioinformatics and Systems Biology, Helmholtz Centre, Munich, BAV, Germany), and Theo Geisel (Nonlinear Dynamics, Max Planck Instittute for Dynamics and Self-Organization, Goettingen, Nds, Germany)

Although human musical performances represent one of the most valuable achievements of mankind, the best musicians perform imperfectly. Musical rhythms are not entirely accurate and thus inevitably deviate from the ideal beat pattern. Nevertheless, computer generated perfect beat patterns are frequently devalued by listeners due to a perceived lack of human touch. Professional audio editing software therefore offers a humanizing feature which artificially generates rhythmic fluctuations. However, the built-in humanizing units are essentially random number generators producing only simple uncorrelated fluctuations. Here, for the first time, we establish long-range fluctuations as an inevitable natural companion of both simple and complex human rhythmic performances [1,2]. Moreover, we demonstrate that listeners strongly prefer long-range correlated fluctuations in musical rhythms. Thus, the favorable fluctuation type for humanizing interbeat intervals coincides with the one generically inherent in human musical performances. [1] HH et al., PLoS ONE,6,e26457 (2011). [2] Physics Today, invited article, submitted (2012).

8:50

4aMUa2. Human body rhythms motion analogy in music sound. Alexander Ekimov (National Center for Physical Acoustics, The University of Mississippi, 1 Coliseum Drive, University, MS 38677, aekimov@olemiss.edu)

The universal algorithm developed for searching periodic and quasiperiodic rhythms in different type of signals [JASA 129(3)] was applied for a processing a few musical sound files and the results were reported on the ASA and other conferences and published in the POMA 14 (2011) article. Originally this algorithm was developed for finding human body motion rhythms in signals of different security systems. A preliminary conclusion from usage of this algorithm for a few music sound files founded rhythms in music files corresponded to rhythms of human regular body mechanical motion. Its appears that the musicians body parts motions, due to playing music can be found in rhythms of the playing music, which create an impression for the audience. These rhythms in analyzed music sound files are corresponding to mechanical human body movements due to walking or running also. More music file (including vocal) analysis with this rhythm algorithm and the results corresponding to rhythms of the human body motion are presented. This work was supported by the National Center for Physical Acoustics (NCPA) of the University of Mississippi.

9:10

4aMUa3. Heartbeat and ornaments: More technical secrets of swing rhythm. Ken Lindsay (Software Engineering, Tinmap, 180 Ohio St, Ashland, OR 97520, ken@tinmap.com) and Pete Nordquist (Computer Science, Southern Oregon University, Ashland, OR)

We previously demonstrated technically precise methods characterizing various types of Swing style in music. Our primary tool is "diffdot" notation showing, in graphical form, the exact timing relationships between various musical notes that create a specific musical phrase. We have shown several common and obvious details of Swing, all based on time variations from a uniform square grid (Classical Mozart/Bach). Micro-timing variations are generally recognized as being essential to Swing. There may be other elements which define Swing feeling but we have focused on micro-timing details. These are a fruitful source for technical analysis of Swing styles.Triplet subdivision is often associated with Swing – Jazz, Blues – but triplets are neither necessary nor sufficient to distinguish a performance as "Swing" versus "Straight" time. One seemingly universal detail of Swing is an asymmetrical "pulse" or basic beat, e.g. on the downbeat of every measure, or the one and three beat in a 4/4 piece. The time between the two heartbeat notes as they repeat their cycle is not equal. This gives rise to unmistaken Swing. Other non-uniform but precisely repeated timing patterns characterize Swing at hierarchical levels different from pulse. These we call "ornaments" in keeping with common musical jargon.

9:30

4aMUa4. Identifying highly rhythmic stretches of speech with envelope-driven resonance analysis. Sam Tilsen (Linguistics, Cornell University, 203 Morrill Hall, Ithaca, NY 14853, tilsen@cornell.edu)

This paper proposes envelope-driven resonance analysis as a technique for characterizing rhythmicity in speech, emphasizing the degree to which a brief stretch of speech creates a rhythmic expectancy. Most approaches to characterizing the rhythm of speech have utilized measurements derived from the durations of linguistically relevant units such as feet, syllables, or vocalic/consonantal intervals. Recently, alternative approaches have been developed which are based upon the amplitude envelope of the speech waveform after the waveform has been filtered to emphasize low-frequency oscillations associated with alternations between vowels and consonants. These approaches include spectral analysis and empirical mode decomposition of the envelope. The method explored here is resonance analysis, which utilizes a bank of resonators that differ in their characteristic resonant frequencies. The resonators are 2nd order dynamical systems analogous to driven, damped springs. The powers of the resonator amplitudes are analyzed during and subsequent to excitation by a speech amplitude envelope. The power and frequency distribution of the resonant response is used to identify highly rhythmic stretches of speech and characterize their spectral properties.

9:50

4aMUa5. Using resonance to study the deterioration of the pulse percept in jittered sequences. Marc J. Velasco and Edward W. Large (Center for Complex Systems and Brain Sciences, Florida Atlantic University, 777 Glades Rd, Boca Raton, FL 33432, velasco@ccs.fau.edu)

Studies of pulse perception in rhythms often ask what periodicity describes the pulse, e.g., tempo identification. In studies of pulse attribution, irregular rhythmic sequences are rated for the degree to which a pulse percept is elicited, if at all. Here, we investigate how a resonance approach to pulse perception may explain the reduction in pulse attribution ratings for jittered sequences while also predicting perceived tempo. We use a signal processing approach to predict perceptual ratings and behavioral performance measures (i.e., tapping data). Measures of resonance are evaluated using both FFT and a network of neural oscillators. The stimuli were isochronous sequences modified with varying levels of pseudorandom Kolakoski jitter. In separate blocks, participants were asked to provide pulse attribution judgments and to tap at the pulse rate. As levels of jitter increased, pulse attribution ratings decreased and participants tapped periodically at the mean sequence rate. At certain high levels of jitter, pulse attribution ratings increased and participants entrained at a new tapping rate. Resonance measures account for both mean tapping rate and pulse attribution ratings, suggesting that these two behavioral measures may be different aspects of the same resonant phenomenon.

10:10–10:25 Break

10:25

4aMUa6. Rhythm and meter in 21st century music theory. Justin London (Music, Carleton College, One North College St., Northfield, MN 55057, jlondon@carleton.edu)

Theories of rhythm in western music go back to Aristoxenus (335 BC) and have continued unabated to the present day. Yet while music theoretic discussions of melody and harmony since Pythagoras have often looked to mathematics and acoustics, only recently has music theory availed itself of research in acoustics, psychoacoustics, and auditory psychology. The central question for a theory of musical rhythm is "what makes something regular enough to be considered a rhythm?" Answering this question requires not only a knowledge of music in a range of musical styles and cultures, but also understanding of our basic psychological capacities for temporal perception and discrimination, as well as our perceptual biases and habits. A brief outline of recent theories of rhythm and meter that draw upon these domains will be presented, with an emphasis on musical meter as kind of entrainment, that is, a synchronization of our attending and/or sensorimotor behaviors to external periodicities in a particular temporal range.

10:45

4aMUa7. Cross-cultural concepts and approaches in musical rhythm. Rohan Krishnamurthy (Musicology, Eastman School of Music, Rochester, NY 14604, rohan.krishnamurthy@rochester.edu)

Western (written, notated) and Indian (oral, unnotated) systems of musical rhythm will be analyzed from theoretical and performance perspectives. Rhythmic concepts and parameters such as meter and tala (rhythmic cycles with constant duration and tempo), tuplet and nadai subdivisions of an underlying pulse, and accelerando (gradual accelerations in musical tempo) and decelerando (gradual decelerations in musical tempo) will be defined in a cross-cultural context. These systems of understanding temporal flow have wide-ranging implications on musical form, style and aesthetics, and artistic freedom. The corporeal or physical dimension of musical rhythm, resulting from instrumental techniques, vocalizations of rhythms, and physical systems of constructing and maintaining temporal flow such as tala visualizations and ensemble conducting, will also be considered. The presentation will include live, interactive musical demonstrations and will be followed by a performance on the mridangam, the primary percussion instrument from South India.

Contributed Paper

11:05

4aMUa8. Analysis of rhythm performance strategies on the Indian tabla as a function of tempo. Punita G. Singh (Sound Sense, 16 Gauri Apartments, 3 Rajesh Pilot Lane, New Delhi 110011, India, punita@gmail.com)

In north Indian classical music, the range of tempi can extend from the ultra-slow 'vilambit' at less than a beat every 5 seconds to the super-fast 'drut' at over 10 beats per second. To hold a rhythm at these speeds and generate a perceptible metrical structure, performers routinely alter playing strategies that derive from neurophysiological and psychoacoustical considerations. At slow speeds, theoretically silent intervals are in practice punctuated by filler sounds to maintain perceptual connectivity. At high speeds, an interesting phenomenon is observed as compound sounds or 'bols' segregate into their simpler components, forming auditory streams of acoustically similar sounds. Compound bols such as 'dha' break up into the tonal 'ta' and the noisy 'ghe', with the sequence of rapidly recurring 'ghe' sounds forming a noise band that could potentially mask tonal accent markers. To avoid this, performers routinely drop out the 'ghe' sounds at high speeds at metrically unimportant points in the sequence, while retaining them at points that would mark accents. These playing strategies are useful in providing mental and physical relief to performers in maintenance of a steady rhythm across such a vast range of tempi while also preserving the rhythmic integrity of the music for listeners.

THURSDAY MORNING, 25 OCTOBER 2012 ANDY KIRK A/B, 11:30 A.M. TO 12:00 NOON

Session 4aMUb

Musical Acoustics: Demonstration Performance on the Mridangam by Rohan Krishnamurthy

James P. Cottingham, Chair
Physics, Coe College, Cedar Rapids, IA 52402

Rohan Krishnamurthy will present a percussion solo on the ancient South Indian pitched drum, the mridangam. The performance will showcase the lively and complex rhythmic nuances of Indian classical music and involve interactive audience participation.

THURSDAY MORNING, 25 OCTOBER 2012 TRIANON C/D, 8:20 A.M. TO 11:25 A.M.

Session 4aNS

Noise, Architectural Acoustics, and ASA Committee on Standards: Ongoing Developments in Classroom Acoustics—Theory and Practice in 2012, and Field Reports of Efforts to Implement Good Classroom Acoustics I

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

Louis C. Sutherland, Cochair
lcs-acoustics, 5701 Crestridge Rd., Rancho Palos Verdes, CA 90275

Chair's Introduction—8:20

Invited Papers

8:30

4aNS1. Classroom acoustics 2012: Recent developments, current issues, and field reports. David Lubman (DL-Acoustics, Westminster, CA) and Louis C. Sutherland (LCS-Acoustics, 5701 Crestridge Rd, Apt 243, Rancho Palos Verdes, CA 90275, lou-sutherland@juno.com)

This introductory paper provides an overview of the papers in this session. It showcases important findings of the UK's Essex Study by David Canning & Adrian James (2012) which confirms large listening benefits for reducing reverberation times (RT) to 0.4 sec or less. The Essex study also found a marked drop in LA90 for occupied classrooms when RT was halved. This introductory paper also briefly reviews the Acoustical Society of America's initial actions leading to development of the influential ANSI standards on classroom acoustics (S12.60 - 2010/Parts1 and 2), and subsequent outreach actions, including publication of Classroom Acoustics booklets. (Two new booklets, one aimed at Educators and the other aimed at Architects, are being prepared for publication.) Also reviewed is the ongoing struggle to incorporate meaningful noise and reverberation time criteria into design guidelines for the California Collaborative for High Performance Schools, Los Angeles Unified School District, and LEED for Schools. Finally, it shows that noise transients occurring during classroom noise measurements can make quiet classrooms seem misleadingly noisy.

8:50

4aNS2. Essex experimental study: The impact of reverberation time on working classrooms. David Canning (UCL, Gower Street, London WC1E 6BT, United Kingdom, canningd@gmail.com), Adrian James (Adrian James Acoustics, Norwich, United Kingdom), and Bridget M. Shields (Urban Engineering, London South Bank University, London, United Kingdom)

There has been considerable debate regarding the value of adding acoustic treatment to refurbished classrooms. Is there any demonstrable benefit in reducing reverberation time in secondary schools to below 0.8s? This study aimed to examine the impact of reverberation time on working classrooms. Four similar classrooms with RT $>$ 0.9s were selected for the study. Three rooms were treated with visually similar acoustically absorbent materials to reduce the RT to between 0.3 and 0.8s, the fourth room being left as a control. Over a period of six months the treatments were changed so that all class/teacher combinations experienced the different acoustic environments, while remaining blind to the condition. Ten teachers and 400 children including 17 hearing impaired children were involved in the study. Extensive objective and qualitative (interview and questionnaire) data were collected throughout the project. Analysis of the impact of room acoustics on classroom noise was conductedblind to the acoustic condition. Results demonstrate that RT has a significant impact on classroom noise levels and occupant behaviour. Reductionof reverberation time from 0.8 to 0.4s brought a reduction of 9 dB in LA90 as against the expected 3dB reduction. Qualitative data supports the beneficial impact on the classroom experience.

9:10

4aNS3. Impact and revision of UK legislation on school acoustics. Bridget M. Shield and Robert Conetta (Urban Engineering, London South Bank University, Borough Road, London SE1 7JQ, United Kingdom, shieldbm@lsbu.ac.uk)

Since 2003 new school buildings in England and Wales have been subject to Building Regulations which impose a legal requirement for spaces in schools to meet acoustic performance criteria for ambient noise levels, reverberation times and sound insulation. The criteria are specified in the Department of Education publication 'Building Bulletin 93' (BB93). In 2008 it was agreed that BB93 would be updated. The Labour government endorsed the need for good acoustic design of schools and agreed to a minor revision of the legislation. However, the new government elected in 2010 recommended the removal of legislation on school acoustics, in order to reduce the cost of new school buildings. The acoustics community in the UK successfully lobbied the government to keep the legislation and it has been agreed that the acoustic regulations relating to the performance of a building in use will be retained. BB93 is currently (June 2012) being redrafted and the acoustic performance specifications revised. This paper will use the results of a recent large scale survey of the acoustics of secondary schools in the UK to examine the impact of BB93 on school design over the past 10 years, and will discuss the current revision of the legislation.

9:30

4aNS4. Effects of noise in high schools on pupils' perceptions and performance. Julie E. Dockrell, Daniel Connolly (Psychology and Special Needs, Institute of Education, London University, London, United Kingdom), Charles Mydlarz (School of Computing, Science and Engineering, University of Salford, Manchester, United Kingdom), Robert Conetta, Bridget M. Shield (Urban Engineering, London South Bank University, Borough Road, London SE1 7JQ, United Kingdom, shieldbm@lsbu.ac.uk), and Trevor J. Cox (School of Computing, Science and Engineering, University of Salford, Manchester, United Kingdom)

A recent project has investigated acoustical conditions in secondary (high) schools, and examined the effects of a poor acoustic environment on teaching and learning of 11- to 16-year-olds. Around 2600 pupils from suburban secondary schools in England responded to an online questionnaire concerning the acoustic environment in their schools. The questionnaire data highlighted the differential effects of noise reported by more vulnerable learners. A repeated measures experimental study involving 572 pupils examined reading performance under two different classroom noise simulations. Results revealed a complex pattern reflecting noise levels, time of testing and measure of reading performance used. Reading text while exposed to classroom noise of 70 dB resulted in quicker reading but less accuracy in measures of reading comprehension compared with performance in 50 dB. The data further suggested that the pupils were not processing the text as deeply as was evident from their reduced lexical learning. There were also interactions with time of testing highlighting the importance of examining the effects of chronic exposure in addition to single session experimental testing. The test results show that capturing the effects of noise on pupils' learning in realistic classroom environments raises a number of methodological and analytical problems.

9:50

4aNS5. Classroom acoustics and beyond: Soundscapes of school days. Jeff Crukley (London Hearing Centre, 1843 Bayswater Crescent, London, ON N6G 5N1, Canada, jcrukley@gmail.com)

Moving beyond traditional measures of classroom acoustics, in this presentation I propose a novel approach that addresses the dynamic nature of the school-day soundscape. In addition to noise floor and reverberation measures, I suggest that the use of dosimetry and observation of children's acoustic environments and situations can provide a more realistic representation of children's listening needs and the contexts of potential challenges. Cohorts of daycare, elementary, and high school students were shadowed by a researcher wearing a dosimeter and recording observational data. Detailed tracings of the sound levels, the types and sources of sounds, and classifications of the acoustic situations will be presented. Results demonstrated a wide range of listening environments, goals, and situations across all three cohorts of students. Sample recordings from the school day soundscapes will be presented. The implications of these results for how we think about and study classroom acoustics will be discussed.

10:10–10:30 Break

10:30

4aNS6. Ongoing developments in classroom acoustic theory and practice in 2012, and reports on efforts to implement good classroom acoustics. Pamela Brown and Mary Crouse (David H. Sutherland & Co., Inc., 2803 NE 40th, Tigard, Portland, OR 97220, mcrouse@comcast.net)

We live in a time of increasingly loud competing sounds and hearing loss is the number one disability in the world. Diverse populations of school children are especially vulnerable. The result is a degradation of the child's academic achievement. New classrooms, built everyday, often incorporate acoustical barriers which limit students' achievements. Overcoming these barriers involves funding constraints, construction timelines and lack of support which requires advocacy from parents, school boards, and design teams. This advocacy should include the ANSI Classroom Acoustics standards and an acoustical assessment of existing classrooms. Complex classroom acoustics challenges may include reduction of noise radiated by HVAC systems, improved acoustic treatment of external walls to minimize exterior noise and acoustic design of walls between adjacent noisy classrooms. Next steps for schools should be to retain an architect and/or an acoustical engineer for remodels and new school construction who are well versed in acoustics for educational settings and noise control. A booklet covering these issues, and designed as a practical guide for educators not versed in acoustics, is in preparation by the Acoustical Society of America.

10:50

4aNS7. Creation of an architects' companion booklet for ANSI 12.60 American National Classroom Acoustics Standard. David S. Woolworth (Oxford Acoustics, 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and Peter Phinney (Bryant Palmer Soto, Inc., Torrence, CA)

This paper outlines the process of collaboration between an architect and acoustician to produce a document that translates the fundamental objectives and ideas of ANSI 12.60 from the semi-archaic language of an acoustics standard to a simple, useful reference for all stripes of architects. Included will be the paradigm of the approach, definition of scope presented, order of presentation, and methods of presentation.

Contributed Paper

11:10

4aNS8. Acoustic design of a new elementary school to meet high performance prerequisites using a school districts base design: Predictions and results from commissioning. Steve Pettyjohn (The Acoustics & Vibration Group, Inc., 5700 Broadway, Sacramento, CA 95820, spettyjohn@acousticsandvibration.com)

An architectural firm was selected to design a new elementary school using the school district's standard building, but with modifications to meet the prerequisites of the Collaborative for Performance Schools (CHPS). Two acoustic prerequisites are a part of the CHPS programs including a background limit of 45 dB(A) and a reverberation time of 0.6 seconds. A 2-story design forms the basis of design. First tests were done at an existing elementary school with the same design. Acoustical recommendations for wall designs, room finishes and HVAC design were incorporated into the design and construction of the new school. The school was not near significant transportation noise sources. After construction was mostly complete, tests were done to learn the sound transmission loss of walls and floor/ceiling systems. Reverberation time tests and background sound levels were measured after construction was complete. Background sound met design goals in all but one space except for the sound generated by a wind turbine mounted on one end of the buildings. This was added by the schools Principal during the latter part of construction without consulting everyone. This proved to be a significant source that had to be removed.

THURSDAY MORNING, 25 OCTOBER 2012 TRIANON A, 8:00 A.M. TO 11:15 A.M.

Session 4aPA

Physical Acoustics and Noise: Infrasound I

Roger M. Waxler, Chair
NCPA, University of Mississippi, University, MS 38677

Invited Papers

8:00

4aPA1. Coherent ambient infrasound recorded by the International Monitoring System. Robin S. Matoza (Institute of Geophysics and Planetary Physics, Scripps Institution of Oceanography, UC San Diego, IGPP 0225, La Jolla, CA 92093-0225, rmatoza@ucsd.edu), Matthieu Landes, Alexis Le Pichon (DAM/DIF, CEA, Arpajon, France), Lars Ceranna (BGR, Hannover, Germany), and David Brown (IDC, Comprehensive Nuclear Test-Ban Treaty Organization (CTBTO), Vienna, Austria)

Ambient noise recorded by the International Monitoring System (IMS) infrasound network includes incoherent wind noise and coherent infrasonic signals; both affect detection of signals of interest. We present summary statistics of coherent infrasound recorded by the IMS network. We have performed systematic broadband (0.01-5 Hz) array processing of the IMS historical dataset (39 stations

from 2005 to 2010) using an implementation of the Progressive Multi-Channel Correlation (PMCC) algorithm in log-frequency space. From these results, we estimate multi-year 10th, 50th, and 90th percentiles of the rms pressure of coherent signals in 15 frequency bands for each station. We compare the resulting coherent noise models in the 15 frequency bands with raw power spectral density noise models, which inherently include both incoherent and coherent noise. We show that IMS arrays consistently record coherent ambient infrasound across the broad frequency range from 0.01 to 5 Hz when wind-noise levels permit. Multi-year averaging of PMCC detection bulletins emphasizes continuous signals such as oceanic microbaroms, as well as persistent transient signals such as repetitive volcanic, surf, or anthropogenic activity (e.g., mining or industrial activity).

8:20

4aPA2. Modeling the generation of infrasound from earthquakes. Stephen Arrowsmith (Geophysics Group, Los Alamos National Laboratory, 1711 Second Street, Santa Fe, NM 87505, sarrowsmith@gmail.com), Relu Burlacu, Kristine Pankow (Seismograph Stations, University of Utah, Salt Lake City, UT), Brian Stump (Huffington Department of Earth Sciences, Southern Methodist University, Dallas, TX), Richard Stead, Rod Whitaker (Geophysics Group, Los Alamos National Laboratory, Los Alamos, NM), and Chris Hayward (Huffington Department of Earth Sciences, Southern Methodist University, Dallas, TX)

Earthquakes can generate complex seismo-acoustic wavefields, consisting of seismic waves, epicenter-coupled infrasound, and secondary infrasound. We report on the development of a numerical seismo-acoustic model for the generation of infrasound from earthquakes. We model the generation of seismic waves using a 3D finite difference algorithm that accounts for the earthquake moment tensor, source time function, depth, and local geology. The resultant acceleration-time histories (on a 2D grid at the surface) provide the initial conditions for modeling the near-field infrasonic pressure wave using the Rayleigh integral. Finally, we propagate the near-field source pressure through the Ground-to-Space atmospheric model using a time-domain parabolic equation technique. The modeling is applied to an earthquake of MW 4.6, that occurred on January 3, 2011 in Circleville, Utah; the ensuing predictions are in good agreement with observations made at the Utah network of infrasonic arrays, which are unique and indicate that the signals recorded at 6 arrays are from the epicentral region. These results suggest that measured infrasound from the Circleville earthquake is consistent with the generation of infrasound from body waves in the epicentral region.

8:40

4aPA3. The variability in infrasound observations from stratospheric returns. Läslo Evers and Pieter Smets (KNMI, PO Box 201, De Bilt 3730 AE, Netherlands, evers@knmi.nl)

Long range infrasound propagation depends on the wind and temperature around the stratopause (alt. 50 km). There is a seasonal change in the wind direction around the equinoxes. In summer, the wind and temperature structure of the stratosphere is stable. In winter, however, planetary waves in the troposphere can travel into the stratosphere and disturb the mean flow. This mean flow is most pronounced in the stratospheric surf zone from 20N (20S) to 60N (60S). One of the most dramatic events in the stratosphere is a Sudden Stratospheric Warming (SSW) during the winter. These occur every winter on the Northern Hemisphere as minor Warmings with a major SSW each other year. SSWs have a strong influence on infrasound propagation due to the large change in temperature and possible reversal of the wind. Therefore, SSWs are important to consider in relation to, e.g., regional and global monitoring with infrasound for verification purposes or other strategic deployments. In this presentation, the detectability of infrasound will be considered as a function of the state of the stratosphere. Variations in strength of the circumpolar vortex (around the stratopause) and temperature changes will give rise to specific propagation conditions which can often not be foreseen.

9:00

4aPA4. Anomalous transmission of infrasound through air-water and air-ground interfaces. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Research Laboratory, Physical Sciences Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

Sound speed and especially mass density exhibit large relative changes at gas-liquid and gas-solid interfaces. Sound transmission through an interface with a strong impedance contrast is normally very weak. However, diffraction effects can lead to the phenomenon of anomalous transparency of gas-liquid or gas-solid interfaces, where most of the acoustic power generated by a compact, low-frequency source located within the liquid or within the solid is radiated into the gas. Contrary to the conventional wisdom based on ray-theoretical predictions and observations at higher frequencies, infrasonic energy from compact waterborne and underground sources can be effectively transmitted into air. This paper reviews the theory and emerging experimental evidence of the anomalous transparency. Physical mechanisms responsible for enhanced sound transmission at low frequencies are discussed. The phenomenon of anomalous transparency can have significant implications, in particular, for localization of buried objects and for acoustic monitoring, detection, and classification of powerful underwater and underground explosions for the purposes of the Comprehensive Nuclear-Test-Ban Treaty.

9:20

4aPA5. Observation of the Young-Bedard Effect during the 2010 and 2011 Atlantic Hurricane Seasons. Philp Blom, Roger Waxler, William Garth Frazier, and Carrick Talmadge (National Center for Physical Acoustics, University of Mississippi, 1 Coliseum Dr, University, MS 38677, psblom@olemiss.edu)

Infrasonic acoustic energy is known to be generated during the collision of counter propagating ocean surface waves of like periods. The acoustic signals produced by such collisions are known as microbaroms. One significant source of microbarom radiation is the interaction of waves produced by large maritime storms with the background ocean swell. The region in which the microbaroms associated with a large storm are produced tends to be hundreds of kilometers from the eye of the storm. It was suggested by Young and Bedard that, when observed along propagation paths that pass through the storm, the microbarom signal can be severely refracted by the storm itself. Such refraction has been observed in data from the 2010 and 2011 Atlantic hurricane seasons. A data processing algorithm has been developed and implemented using the Multiple Signal Classification (MUSIC) spatial spectra and Akaike Information Criterion. The results of this analysis will be presented and compared with predictions of the refraction using a geometric acoustics propagation model.

9:40

4aPA6. Exploiting correlation in wind noise for enhanced detection of transient acoustic signals. William G. Frazier (NCPA, University of Mississippi, 1 Coliseum Dr, University, MS 38677, frazier@olemiss.edu)

Wind noise presents significant difficulties when trying to detect transient acoustic signals. The most common approach to enhancing signal detection when the signal-to-noise ratio is low due to wind noise is to utilize mechanical windscreens, large number of widely spaced microphones or a combination of both. Results from recent experimental investigations and algorithm developments are presented that demonstrate a alternative method for improving detection of transients that utilizes only a few closely spaced microphones and a unique processing technique that explicitly exploits the correlation among wind noise induced pressure fluctuations.

10:00–10:10 Break

10:10

4aPA7. Validating infrasound sensor performance: Requirements, specifications, and calibration. Darren M. Hart (Ground Based Nuclear Explosion Monitoring R & D, Sandia National Lab, PO Box 5800, Mail Stop 0404, Albuquerque, NM 87109, dhart@sandia.gov), Rod Whitaker (Earth and Environmental Science, Los Alamos National Lab, Los Alamos, NM), and Harold Parks (Primary Standards Laboratory, Sandia National Lab, Albuquerque, NM)

The Ground-Based Nuclear Explosion Monitoring Research and Development (GNEM R&D) program at Sandia National Laboratories (SNL) is regarded as a primary center for unbiased expertise in testing and evaluation of geophysical sensors and instrumentation for nuclear explosion monitoring. In the area of Infrasound sensor evaluation, Sandia relies on the “comparison calibration” technique to derive the characteristics of a new sensor under evaluation relative to a standard reference infrasound sensor. The traceability of our technique to a primary standard is partially dependent on the infrasound calibration chamber operated by a similar program group at Los Alamos National Laboratory (LANL). Previous work by LANL and the SNL Primary Standards Laboratory was able to determine the LANL chamber pistonphone output pressure level to within 5% uncertainty including dimensional measurements and careful analysis of the error budget. Over the past several years, the staff at LANL and the SNL Facility for Acceptance, Calibration and Testing (FACT) site has been developing a methodology for the systematic evaluation of infrasound sensors. That evaluation involves making a set of measurements that follow a prescribed set of procedures, allowing traceability to a primary standard for amplitude. Examples of evaluation tests will be shown for monitoring quality infrasound sensors.

Contributed Papers

10:30

4aPA8. Noise reduction optimization of wind fences. JohnPaul R. Abbott, Richard Raspet, and Jeremy Webster (Natl. Center for Physical Acoustics–Dept. of Phys. and Astr., The University of Mississippi, 1 Coliseum Dr, University, MS 38677, johnpaul.abbott@gmail.com)

An earlier paper [J. Acoust. Soc. Am. 129, 2445 (2011)] described an investigation on the optimization of a large wind screen for infrasonic noise reduction. This wind screen is a circular variable porous enclosure 3 m high and 5 m in diameter consisting of ten chain link fence panels about 3 m high and 1.5 m wide, with removable vinyl privacy slats, an open top, and a 0.1 m bottom gap. That paper reported on the noise reduction for a microphone set at the center of the enclosure relative to another set outside the enclosure as the screen’s porosity was varied. Both microphones were mounted under porous foam flush to the ground. It was shown that the best reductions occurred at intermediate porosities, with reductions of 6 dB or greater between 0.6 -10 Hz, with max reductions about 13-15 dB. The current paper will report on the effect of further optimization techniques—sealing off the bottom gap, adding a roof, and placing a small porous dome over the enclosed field microphone. Of these techniques the addition of the dome was most effective, with noise reductions of 6 dB or greater between 0.3-10 Hz, with max reductions about 20-23 dB.

10:45

4aPA9. Uncertainty associated with *in-situ* frequency-response estimation by reference-sensor comparison at infrasound monitoring sites. Thomas B. Gabrielson (Applied Research Laboratory, Penn State University, PO Box 30, State College, PA 16804, tbg3@psu.edu)

In-situ measurement of the frequency-response of infrasound array elements has proven to be a useful tool in the assessment of element performance. In order to transition to a true calibration process, the uncertainties inherent in the method must be determined. It is critically important to distinguish between bias errors and random errors and to recognize that the ambient pressure fluctuations are typically not stationary in a statistical sense. The time evolution of the cross-spectrum is particularly useful for identifying non-stationary behavior and for isolating high-quality data intervals. Three important cases are tractable: high coherence between the reference sensor and the infrasound element; low-to-moderate coherence resulting from uncorrelated noise in one channel; and moderate coherence resulting from uncorrelated noise in both channels. For a fixed number of averages, the confidence limits for the frequency-response estimate are often considerably tighter than the corresponding limits for the estimated spectral densities.

11:00

4aPA10. Direct measurement of the acoustical impedance of wind-noise-reduction pipe systems. Thomas B. Gabrielson and Matthew Poese (Applied Research Laboratory, Penn State University, PO Box 30, State College, PA 16804, tbg3@psu.edu)

Wind-noise-reduction systems for infrasound monitoring stations often take the form of networks of pipes and cavities. The acoustical response of these wind-noise-reduction systems can be determined using ambient noise and comparison to a reference sensor. Faults in these systems can sometimes be detected by such response measurements; however, identification and localization of a fault is more challenging. Another approach for performance assessment is to measure the acoustical impedance at accessible points in the pipe network. This approach has the potential for high signal-to-noise ratio, less dependence on atmospheric conditions, and the ability to isolate sub-sections of the network. A portable apparatus has been designed for field measurement of acoustical impedance. The impedance apparatus generates a controlled volume velocity and measures acoustic pressure at the driving point.

Session 4aPP

Psychological and Physiological Acoustics: Physiology and Perception (Poster Session)

Gabriel A. Bargen, Chair
Communication Sciences and Disorders, Idaho State University, Meridian, ID 83642

Contributed Papers

All posters will be on display from 8:30 a.m. to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:30 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m.

4aPP1. Auditory brainstem responses evoked by chirp and click stimuli in children. Gabriel A. Bargen (Meridian-Health Science Center, Idaho State University, 1311 E. Central Drive, Meridian, ID 83642, barggabr@isu.edu)

The chirp-evoked auditory brainstem response (ABR) has been found to be a more synchronous response in adults than the click-evoked ABR with more areas of the cochlea contributing to the compound ABR. ABRs evoked using delayed-model chirp stimuli have shown to compensate for the temporal dispersion of the cochlea and result in larger wave V amplitudes and better overall morphology when compared to click-evoked ABRs. To date, published research has only included adult subjects with the majority of studies completed on subjects with normal hearing. This study compares the chirp-evoked ABR to the click-evoked ABR in children to determine if the chirp-evoked stimulus is more efficient than the click-stimulus which is currently used in most newborn hearing screening protocols and pediatric diagnostic ABR evaluations. Subjects from birth to eight years of age, with normal and abnormal hearing, participated in this study. This presentation will include preliminary study findings.

4aPP2. Effectiveness of steady versus varied-color/varied-pattern visual tasks during acquisition of late auditory evoked potentials. Charles G. Marx and Edward L. Goshorn (Speech and Hearing Sciences, University of Southern Mississippi, Psychoacoustics Research Laboratory, Hattiesburg, MS 39401, edward.goshorn@usm.edu)

Instructions for late auditory evoked potential (LAEP) testing include a need for the subject to remain alert (not go to sleep). Previous studies show an inverse relationship between alertness level and waveform morphology. Thus, a need exists to maintain alertness during LAEP testing. If not maintained, a wide range of alertness, and thus waveform morphology, may exist from one run to the next. Therefore, if alertness level is not controlled, any variations in waveform morphology may be due to variations in alertness rather than auditory system integrity. Previous investigators have implemented visual tasks consisting of still or action images in an attempt to maintain alertness. In these visual tasks, a subject is typically instructed to attend to a video screen during LAEP testing. This project investigated the effectiveness of two visual task screens: unvaried blue with no pattern, versus varied colors-patterns occurring in 1-3 second random intervals. LAEPs were gathered on twenty-five young adult subjects who were instructed to attend to a video display of one of the screens during LAEP testing. Six replicates were obtained for each screen in counter-balanced order. Results showed no significant ($p>.05$) differences in mean P1 or P2 latency or amplitude for the two screens.

4aPP3. Links between mismatch negativity responses and speech intelligibility in noise. Tess K. Koerner, Yang Zhang, and Peggy Nelson (Department of Speech-Language-Hearing Sciences, University of Minnesota, Minneapolis, MN 55408, koern030@umn.edu)

Research has shown that the amplitude and latency of neural responses to passive mismatch negativity (MMN) tasks are affected by noise (Billings et al., 2010). Further studies have revealed that informational masking noise results in decreased P3 amplitude and increased P3 latency, which correlates with decreased discrimination abilities and reaction time (Bennett et al., 2012). This study aims to further investigate neural processing of speech in differing types of noise by attempting to correlate MMN neural responses to consonant and vowel stimuli with results from behavioral sentence recognition tasks. Preliminary behavioral data indicate that noise conditions significantly compromise the perception of consonant change in an oddball discrimination task. Noise appears to have less of an effect on the perception of vowel change. The MMN data are being collected for the detection of consonant change and vowel change in different noise conditions. The results will be examined to address how well the pre-attentive MMN measures at the phonemic level can predict speech intelligibility at the sentence level using the same noise conditions.

4aPP4. Effect of broadband contralateral noise on distortion product otoacoustic emissions and psychophysical tuning curves. Andrzej Wicher (Institute of Acoustics AMU, Umultowska85, Poznan 61-614, Poland, awaku@amu.edu.pl)

The main purpose of this work was to describe an influence of contralateral stimulation (CS) on distortion products otoacoustic emissions (DPOAEs) and psychophysical tuning curves (PTCs). The fast method for determining PTCs was used in the study. DPOAEs and PTCs were measured in two modes: in the presence or absence of CS. The CS was a broadband noise at a level of 50 SPL. The primary tones with frequencies f_1 and f_2, ($f_2/f_1 = 1.21$) were presented at levels of $L_1 = 60$ dB SPL, and $L_2 = 50$ dB SPL. A pulsed sinusoidal signal at a sensation level (SL) of 10 dB was used in the measurements of the PTC. The signal frequency was 1 or 2 kHz. Ten normal-hearing subjects participated in this study. The CS caused a decrease in the level of the DPOAEs (suppression effect) in 90% of cases, in the whole frequency range of f_2 (i.e. from 845 to 6200 Hz). The maximum suppression of the DPOAE level occurs for the f_2 frequency from 1 to 2 kHz. For both signal frequencies the CS significantly reduces the sharpness of the PTCs. The CS has a significant effect on decreasing the quality factor (Q_{10}) of PTCs.

4aPP5. Improving the discriminability of simultaneous auditory alarms using principles of voice-leading found in music. Matthew J. Davis and Nial Klyn (Speech and Hearing Science, The Ohio State University, Columbus, OH 43210, davis.3131@.osu.edu)

Predicting the ability of listeners to discriminate between simultaneous auditory streams is a longstanding challenge in the design of auditory displays. The creation of an efficacious artificial auditory scene can require an immense amount of knowledge about how sounds are heard and interpreted; what is commonly called auditory scene analysis. Fortunately, musicians have been constructing novel auditory scenes with multiple simultaneous streams for many centuries, and the rules governing the composition of Western polyphonic music have even been explicitly codified in a range of techniques referred to as “voice-leading”. These relatively simple but effective rules have the potential to help guide designers of auditory displays by maximizing the distinctions between concurrent signals. An experiment was conducted to measure the discriminability of alarms designed with musical

4a THU. AM

"voice-leading" features as compared with existing alarms from Learjet 31a and Learjet 35 aircraft. Signals designed with the auditory scene synthesis techniques embedded in musical "voice-leading" were found to significantly improve discriminability for up to five simultaneous alarms. By applying these principles to warning signals, this study has sought to implement a system for creating new auditory warnings that contain more efficient differentiating properties and furthermore conform to a more unified stylistic identity.

4aPP6. Effects of listener bias on auditory acuity for aircraft in real-world ambient environments. Matthew J. Davis, Lawrence L. Feth (Speech and Hearing Science, The Ohio State University, Columbus, OH 43210, davis.3131@osu.edu), Michael Spottswood, and John Hall (711 Human Performance Wing, Air Force Research Laboratory, Wright Patterson Air Force Base, OH)

Hoglund et al. (2010) investigated the ability of listeners to detect the presence of aircraft masked by ongoing ambient sounds using a two interval forced choice (2IFC) procedure. They found that the signal-to-noise ratio required for target detection varied across the different types of ambient environments. Recordings of helicopters in flight were used as target signals and maskers were recorded in rural, suburban and urban locations. Their goal was to better approximate real-world conditions. The goal of the current study is to extend those results to include factors that may bias the listener under more realistic conditions. The 2IFC procedure is designed to minimize listener bias; however, real-world listening conditions are more typically one interval situations. The frequency of occurrence of aircraft over-flights and the costs of errors and rewards for correct responses may substantially affect some estimates of listener sensitivity. Work reported here investigated the influence of apriori probability of target occurrence and manipulation of the pay-off matrix on the acuity measures reported by Hoglund, et al., using the same target sounds and environmental maskers. Psychometric functions shifted by ~18 dB as frequency of targets varied from 20% to 80%. ROC curves display the influence of pay-off manipulations.

4aPP7. Modulation difference limen for spectral center-of-gravity signals. Amy E. Stewart, Evelyn M. Hoglund, Yonghee Oh, and Lawrence L. Feth (Speech and Hearing Science, Ohio State University, 110 Pressey Hall, 1070 Carmack Road, Columbus, OH 43210, feth.1@osu.edu)

Auditory processing of the dynamic spectral center-of-gravity (COG) of a pair of amplitude modulated (AM) tones was investigated by comparing the modulation difference limen (DL) for a COG signal to that for a sinusoidally frequency modulated (FM) tone. The center-of-gravity effect refers to the listener's ability to track an amplitude-weighted instantaneous frequency between two tones differing in frequency. To create a dynamic COG, two tones separated in frequency by four ERB were amplitude modulated at the same modulation rate and modulation depth. AM modulators differed only in relative phase. For five normal-hearing listeners, a 2IFC discrimination task was used to determine the DL for frequency deviation across a range of center frequencies, modulation frequencies, and frequency deviations for both FM and COG signals. COG signals were matched to FM signals (same center frequency, modulation frequency, and frequency deviation). Frequency deviation was determined by equating the maximum instantaneous spectral centroid for each signal type. COG DLs were approximately three times larger than the corresponding FM DLs; however, variation with modulation frequency and frequency deviation was similar for the two types of signals. Results indicate comparable auditory processing for the two types of signals.

4aPP8. Temporal weighting for interaural time differences in low-frequency pure tones. Anna C. Diedesch, Jacqueline M. Bibee, and G. Christopher Stecker (Speech & Hearing Sciences, University of Washington, Seattle, WA 98110, diedesch@uw.edu)

In contrast to envelope-based interaural time differences (ITD) at high frequencies, where sound onsets play a dominant role, the reliability and salience fine-structure ITD at low frequency (<1500 Hz) suggests uniform sensitivity to information across periods of an ongoing stimulus waveform. Several past studies, however, have demonstrated low-frequency ITD thresholds to improve sub-optimally with increasing sound duration [e.g. Houtgast & Plomp 1968, JASA 44:807-12], suggesting that the initial periods of a brief tone play a greater role in ITD processing than do later periods. Here, we measured the temporal profile of ITD sensitivity in pure tones ranging from 250-1000 Hz. Sounds were presented with ITD that either remained fixed over the sound duration (condition RR) or progressed linearly to eliminate the ITD cue from either the beginning (condition 0R) or end (R0) of the sound. Durations varied from 40-640 ms, including 20 ms ramps applied diotically to minimize envelope cues. ITD detection thresholds demonstrated (a) suboptimal improvement with duration and (b) greater sensitivity to ITD available early (R0) rather than late (0R) in the stimulus, a pattern nearly identical to that observed for high-frequency envelope ITD. [Supported by NIH R01 DC011548.]

4aPP9. Novelty detection of covariance among stimulus attributes in auditory perception. Christian Stilp (Department of Psychological and Brain Sciences, University of Louisville, Louisville, KY 40292, christian.stilp@gmail.com) and Keith Kluender (Speech, Language, and Hearing Sciences, Purdue University, West Lafayette, IN)

Novelty detection is characterized by enhanced response to a stimulus with some property changed relative to expected input. Many reports examine sensitivity to deviations in physical acoustic dimensions, patterns, or simple rules, but fail to consider information in higher-order statistical relationships between dimensions. Here we report novelty detection that depends upon encoding of experienced covariance between complex acoustic dimensions (attack/decay, spectral shape.) Here, novelty is defined as violation of experienced covariance between otherwise independent acoustic attributes. Listeners primarily discriminated sound pairs in which attributes supported robust covariance (15 pairs, Consistent condition) and rarely discriminated sounds that violated this redundancy (1 pair, Orthogonal condition) in randomized AXB trials without feedback. Probability of occurrence for Orthogonal trials was minimized by withholding them until the final testing block. Discrimination accuracy for Orthogonal sounds exceeded that for Consistent sounds as well as that for control stimuli absent experienced redundancy between attributes. Increasing Orthogonal trial probability reduces this enhancement, as does acoustic similarity between Consistent and withheld Orthogonal sound pairs. Results parallel novelty detection as measured by stimulus-specific adaptation and mismatch negativity. Implications for high-level auditory perception and organization will be discussed. [Supported by NIDCD.]

4aPP10. Using channel-specific models to detect and remove reverberation in cochlear implants. Jill M. Desmond, Chandra S. Throckmorton, and Leslie M. Collins (Department of Electrical and Computer Engineering, Duke University, Durham, NC 27713, jill.desmond@duke.edu)

Reverberation results in the smearing of both harmonic and temporal elements of speech through self-masking (masking within an individual phoneme) and overlap-masking (masking of one phoneme by a preceding phoneme). Self-masking is responsible for flattening formant transitions, while overlap-masking results in the masking of low-energy consonants by higher-energy vowels. Reverberation effects, especially the flattening of formant transitions, are especially detrimental to cochlear implant listeners because they already have access to only limited spectral and temporal information (Kokkinakis and Loizou, 2011). Efforts to model and correct for reverberation in acoustic listening scenarios can be quite complex, requiring estimation of the room transfer function and localization of the source and receiver. However, due to the limited resolution associated with cochlear implant stimulation, simpler processing for reverberation detection and mitigation may be possible. This study models speech stimuli in a cochlear implant on a per-channel basis both in quiet and in reverberation, where reverberation is characterized by different reverberation times, room dimensions, and source locations. The efficacy of these models for detecting the presence of reverberation and subsequently removing its effects from speech stimuli is assessed. [This work was funded by the National Institutes of Health (NIDCD), R01-DC-007994-04.]

4aPP11. The effect of visual information on speech perception in noise by electroacoustic hearing. Qudsia Tahmina, Moulesh Bhandary, Behnam Azimi, Yi Hu (Electrical Engineering & Computer Science, University of Wisconsin-Milwaukee, 3200 N Cramer St, Milwaukee, WI 53211, huy@uwm.edu), Rene L. Utianski, and Julie Liss (Speech & Hearing Science, Arizona State University, Tempe, AZ)

The addition of amplified low frequency hearing to cochlear implants has been shown to provide substantial performance benefits for cochlear implant (CI) users, particularly in noise. In the current study, we examined the extent to which the presence of visual information (facial movement during speech) augments perception for CI listeners with electroacoustic stimulation (EAS). Two experiments were conducted. In the first one, participants transcribed semantically anomalous phrases in quiet and noise. Intelligibility results showed modest improvements in intelligibility for low and high levels of noise, and dramatic gains (30+ percentage points) in mid-level noise. Error analyses conducted on the transcripts further suggest that the perceptual benefits extended beyond articulatory place information to that of facilitating lexical segmentation. In the second experiment, participants were tested on their recognition of words in sentences corrupted by noise. Results showed significant benefit of hearing aids in EAS patients. However, the benefit of acoustic hearing was not apparent when visual information was available. Our results will provide guidance for auditory rehabilitation strategies in this population.

4aPP12. Optimal categorization of sounds varying on a single dimension. Megan Kittleson (Speech, Language, and Hearing Sciences, University of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, mkittles@email.arizona.edu), Randy L. Diehl (Psychology, University of Texas at Austin, Austin, TX), and Andrew J. Lotto (Speech, Language, and Hearing Sciences, University of Arizona, Tucson, AZ)

Listeners were randomly presented narrow-band filtered noise bursts that varied in filter center frequency from two overlapping, Gaussian-like distributions. Participants mapped these distributions of sounds onto creatures in a video game where they received visual and auditory feedback about their accuracy. Categorization boundaries for each participant were estimated using logistic regression and compared with the optimal boundary from an ideal observer model. The participants appeared to be able to establish near optimal boundaries rapidly and had a remarkable ability to shift these boundaries when the underlying distributions changed - even when these changes were not explicitly signaled. These results suggest that listeners maintain a rather detailed representation of distributional information that is continuously updated during the task. This interpretation is in line with the assumptions underlying many current models of perceptual (statistical) learning in speech perception. However, it is possible to get optimal-like behavior by maintaining a general distributional representation or by using simpler "local" strategies based on only a few of the most recently experienced exemplars. The results will be presented with multiple categorization models, which testify to the difficulty of interpreting claims of distributional learning in categorization. [Work supported by NIH-NIDCD.]

4aPP13. Ictal and interictal changes in auditory processing. David M. Daly (Hugin, Inc, Box 210855, Dallas, TX 75211, openmike@alumni.stanford.edu)

Altered neuronal functioning manifest in seizures can also cause interictal misperceptions. The present case experienced nausea, head-turning, and automatism with loss of consciousness; following this, he could see and hear, but could not speak for up to 30 min. Left hemisphere initiated speech; seizures involved right frontal and anterior temporal areas. He underwent anterior temporal lobe resection, and for the next year, seizures were medically controlled. Then seizures recurred; although he remained conscious, he was often amnestic instead, and, again, post-ictally mute. He underwent resection of right frontal lobe; he recovered over the next year with only prophylactic medication. Patient was tested using pre-recorded sets of GY, BDG, and iIe delivered through headphones [J Neurophysiol. 44:1, 200-22 (1980)]. In the year after first surgery, he classified vowels appropriately, but GY as 'not /ye/' and /ye/, and BDG as /be/ and /de/; right ear and binaural performances were statistically less anomalous than left ear. Following second surgery, he classified GY and BDG appropriately. Left ear performance varied by at most chance from standard; right ear was indistinguishable from the standard ($p < 0.0001$).

THURSDAY MORNING, 25 OCTOBER 2012 BASIE A, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: The Nature of Lexical Representations in the Perception and Production of Speech

Allard Jongman, Cochair
Univ. of Kansas, 1541 Lilac Ln., Lawrence, KS 66045

Joan A. Sereno, Cochair
Linguistics, University of Kansas, Lawrence, KS 66049

Chair's Introduction—8:00

Invited Papers

8:05

4aSC1. The role of phonological alternation in speech production: Evidence from Mandarin tone Sandhi. Stephen Politzer-Ahles and Jie Zhang (Linguistics, University of Kansas, Lawrence, KS 66046, sjpa@ku.edu)

An open question in psycholinguistics is the nature of the phonological representations used during speech production and the processes that are applied to them, particularly between lexical access and articulatory implementation. While phonological theory posits that speakers' grammar includes mechanisms for transforming from input to output forms, whether such mechanisms also are used by the parser during online speech production is unclear. We examined the role of phonological alternations in Mandarin Chinese real and novel compounds using the implicit priming paradigm, which can reveal forms being used prior to articulation. We compared modulations of the implicit priming effect in sets of words that are heterogeneous at the lexical level (where one word has a different lexical tone than the rest) to those in sets that are heterogeneous at the derived level (where a word has the same underlying lexical tone, but that tone surfaces as a different tone because of tone sandhi). Both lexical and derived heterogeneous sets reduced the priming effect, suggesting that phonological alternation was computed abstractly before articulation was initiated. We argue that the progression from underlying phonological representations to articulatory execution may be mediated online by a level at which abstract phonological alternations are processed.

8:25

4aSC2. Discreteness and asymmetry in phonological representations of words. Aditi Lahiri (Centre for Linguistics and Philology, University of Oxford, Walton Street, Oxford OX1 2HG, United Kingdom, aditi.lahiri@ling-phil.ox.ac.uk)

Lexical phonological contrasts are generally binary and abound in asymmetries. For example, vowels can contrast in nasality (oral vs. nasal), but the presence of contrastive nasal vowels implies the presence of oral vowels, and not vice versa. The occurrence of geminates in a language implies the presence of single consonants and therefore, a contrast in consonantal length. Here we address the question of how these asymmetries constrain phonological representations of WORDS in the mental lexicon, and how these constraints affect language processing and change. Various phonological contrasts will be discussed including features, length, and tone, claiming that representations are discrete and asymmetric which in turn lead to asymmetry in processing. Experimental evidence will be presented from behavioural as well as brain imaging studies in Bengali, English, and German.

8:45

4aSC3. From speech signal to phonological features—A long way (60 years and counting). Henning Reetz (Dpt. of Empirical Linguistics, Goethe-University Frankfurt, Georg-Voigt-Str. 6/II, Frankfurt 60325, Germany, reetz@em.uni-frankfurt.de)

When Jakobson, Fant and Halle proposed 1952 their feature system to describe the representation of speech, they wrote: "In decoding a message received (A), the listener operates with the perceptual data (B) which are obtained from the ear responses (C) […] The systematic exploration of the first two of these levels belongs to the future and is an urgent duty." In the last three decades, this approach has been substituted by stochastic modeling to map the speech signal to lexical (word) entries in automatic speech recognition. Although this has lead to working ASR applications, the process of speech understanding by humans is still of 'urgent duty'. The FUL (featural underspecified lexicon) system is one model for this process and this talk will present its methods for mapping the signal onto phonological features, which removes acoustic detail that we assume is irrelevant for (human) speech understanding. The analysis is performed with a high temporal resolution to model the 'online' processing of the human brain and provide redundancy for noisy signals. The ultimate goal is to match the acoustic signal to feature sets that activate possible and suppress improbable word candidates. These features sets themselves are defined by the phonological structure of a language rather than by extensive training with speech material. The presentation includes an online demonstration of the system.

9:05

4aSC4. The exemplar-based lexicon. Keith Johnson (Linguistics, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, keithjohnson@berkeley.edu)

Exemplar-based models of memory have been successful in accounting for a variety of recall and recognition data in general cognitive psychology, and provide an interesting counter-point to other more "standard" models of the mental lexicon. This talk will discuss the ways that the exemplar-based lexicon deals with spoken language variability in auditory word recognition: with emphasis on talker normalization, cross-dialect speech perception, and the recognition of highly variable conversational speech. I will also discuss the use of exemplar-based models in the linguistic theory of sound change, and the relationship between exemplar-based models and neurophonetics. Although the specific modeling strategy employed in exemplar-based modeling is likely over-simplified and wrong in some ways, the success of this type of model indicates that something true is being captured. I will suggest that what makes exemplar-based models useful is that they provide a way for the theorist to include a role for fine phonetic detail in the representation of phonological representations. The ultimate argument is that phonetic memory is gradient as well as categorical, and should be modeled as such.

9:25

4aSC5. Processing pronunciation variants: Rules and representations. Cynthia M. Connine and Stanislav Sajin (Psychology, Binghamton University, PO Box 6000, Binghamton, NY 13902, connine@binghamton.edu)

Both representational and inference rule mechanisms have been proposed for recognizing pronunciation variants. In our work, we have advanced a view for recognizing pronunciation variants in which multiple forms are represented in the lexicon, with non-canonical forms represented based on their frequency of occurrence and canonical forms represented in a privileged (immune to frequency of occurrence) status due to their congruence with orthography. These investigations have focused on variants in which the relevant alternation was word internal (e.g. schwa vowel deletion, flapping and nasal flaps). Other classes of pronunciation variants are formed due to interactions with segmental properties of surrounding words (e.g. place assimilation, fricative assimilation); the processing explanation advanced for such variants has focused on phonological inference rules that recover underlying representations. The current project investigated the relative role of inferential processes and representation in processing variants formed due to interaction at word boundaries (e.g. fricative assimilation).

9:45

4aSC6. The consequences of lexical sensitivity to fine grained detail: Solving the problems of integrating cues, and processing speech in time. Bob McMurray (Psychology, University of Iowa, E11 SSH, Iowa City, IA 52242, bob-mcmurray@uiowa.edu) and Joseph C. Toscano (Beckman Institute for Advanced Science and Technology, University of Illinois, Urbana, IL)

Work on language comprehension is classically divided into two fields. Speech perception asks how listeners cope with variability from factors like talker and coarticulation to compute some phoneme-like unit; and word recognition assumed these units to ask how listeners cope with time and match the input to the lexicon. Evidence that within-category detail affects lexical activation (Andruski, et al., 1994; McMurray, et al., 2002) challenges this view: variability in the input is not "handled" by lower-level processes and instead survives until late in processing. However, the consequences of this have not been fleshed out. This talk begins to explore them using evidence from the eye-tracking paradigms. First, I show how lexical activation/competition processes can help cope with perceptual problems, by integrating acoustic cues that are strung out over time. Next, I examine a fundamental issue in word recognition, temporal order (e.g., distinguishing cat and tack). I present evidence that listeners represent words with little inherent order information, and raise the possibility that fine-grained acoustic detail may serve as a proxy for this. Together these findings suggest that real-time lexical processes may help cope with perceptual ambiguity, and that fine-grained perceptual detail may help listeners cope with the problem of time.

10:05–10:20 Break

10:20

4aSC7. The structure of the lexical network influences lexical processing. Michael S. Vitevitch and Rutherford Goldstein (Psychology, University of Kansas, 1415 Jayhawk Blvd., Lawrence, KS 66045, mvitevit@ku.edu)

Network science is an emerging field that uses computational tools from physics, mathematics, computer science, and other fields to examine the structure of complex systems, and explore how that structure might influence processing. In this approach, words in the mental lexicon can be represented as nodes in a network with links connecting words that are phonologically related to each other. Analyses using the mathematical tools of network science suggest that phonological networks from a variety of languages exhibit the characteristics of small-world networks, and share several other structural features. Studies of small-world networks in other domains have demonstrated that such networks are robust to damage, and can be searched very efficiently. Using conventional psycholinguistic tasks, we examined how certain structural characteristics influence the process of spoken word recognition. The findings from these experiments suggest that the lexicon is structured in a non-arbitrary manner, and that this structure influences lexical processing.

Contributed Papers

10:40

4aSC8. How talker-adaptation helps listeners recognize reduced word-forms. Katja Poellmann (International Max Planck Research School for Language Sciences, P.O. Box 310, Nijmegen 6500 AH, Netherlands, katja.poellmann@mpi.nl), James M. McQueen (Behavioural Science Institute and Donders Institute for Brain, Cognition & Behaviour, Radboud University, Nijmegen, Gelderland, Netherlands), and Holger Mitterer (Max Planck Institute for Psycholinguistics, Nijmegen, Gelderland, Netherlands)

Two eye-tracking experiments tested whether native listeners can adapt to reductions in casual Dutch speech. Listeners were exposed to segmental ([b] > [m]), syllabic (full-vowel-deletion), or no reductions. In a subsequent test phase, all three listener groups were tested on how efficiently they could recognize both types of reduced words. In the first Experiment's exposure phase, the (un)reduced target words were predictable. The segmental reductions were completely consistent (i.e., involved the same input sequences). Learning about them was found to be pattern-specific and generalized in the test phase to new reduced /b/-words. The syllabic reductions were not consistent (i.e., involved variable input sequences). Learning about them was weak and not pattern-specific. Experiment 2 examined effects of word repetition and predictability. The (un-)reduced test words appeared in the exposure phase and were not predictable. There was no evidence of learning for the segmental reductions, probably because they were not predictable during exposure. But there was word-specific learning for the vowel-deleted words. The results suggest that learning about reductions is pattern-specific and generalizes to new words if the input is consistent and predictable. With variable input, there is more likely to be adaptation to a general speaking style and word-specific learning.

10:55

4aSC9. Lexically guided category retuning affects low-level acoustic processing. Eva Reinisch and Lori L. Holt (Psychology, Carnegie Mellon University, 5000 Forbes Avenue, Pittsburgh, PA 15213, evarei@andrew.cmu.edu)

Listeners adapt to non-canonically produced speech by using lexical knowledge to retune phoneme categories. It is unclear, however, whether these retuned categories affect perception at the category level or the signal-to-representation mapping. This was addressed by exploring conditions of cross-speaker generalization of retuned fricatives. During a lexical-decision task, American listeners heard a female Dutch learner of English whose word-final /f/ or /s/ was replaced by an ambiguous sound. At test listeners categorized minimal pairs ending in sounds along [f]-[s] continua spoken by the same female speaker and a new male speaker. Listeners' [f]-[s] categorization for the previously heard speaker shifted as a function of exposure. Generalization to the new speaker was not found when continua between his natural [f]-[s] endpoints were presented. However, listeners did generalize to this voice when presented with only a subset of the male's most [f]-like continuum steps, adjusting the fricative range to match the exposure speaker's, and eliminating a bias toward /s/-responses in the male continua. Listeners thus use short-term acquired knowledge about acoustic properties of phonemes even to interpret upcoming phonemes from previously unheard speakers. Acoustic match, not speaker identity, predicted the results supporting accounts of the effect originating in the early signal-to-representation mapping.

11:10

4aSC10. Lexical effects on the perception of /l/ allophones in English. D. H. Whalen (Speech-Language-Hearing Sciences, City University of New York, 360 Fifth Ave., New York, NY 10016, dwhalen@gc.cuny.edu), Ylana Beller-Marino, Stephanie Kakadelis (Dept. of Linguistics, City University of New York, New York, NY), Katherine M. Dawson (Speech-Language-Hearing Sciences, City University of New York, New York, NY), Catherine T. Best (MARCS Institute, University of Western Sydney, Sydney, NSW, Australia), and Julia R. Irwin (Dept. of Psychology, Southern Connecticut State University, New Haven, CT)

Previous work has shown that perception of allophones of /p/ in English utterances was influenced by lexical status. In nonwords, the aspirated allophone was preferred whether appropriate or not; in words, the appropriate allophone was preferred [Whalen, Best, & Irwin (1997), J. Phonetics, 25, 501-528]. Here, we examined dark and light [l] in English words and nonwords. Dark [l] occurs in syllable codas whereas light [l] occurs in onsets. Items were selected in pairs to balance syllable position in monosyllabic English words and pseudowords, such as "gel"/"ledge", "teal"/"leat", and "beel"/"leeb." Frequency of occurrence for words was also manipulated to explore compatibility with versions of exemplar theory. A phonetician produced two versions of each item, one with a contextually appropriate allophone and one with the inappropriate. Listeners were asked to rate where each acoustically presented item fell on a Likert scale (1-7) between "ideal (native) pronunciation" or "bad (nonnative) pronunciation." Results will be discussed in terms of the underlying representation needed to account for lexical effects in perception. The relationship to phonotactic rules will also be discussed.

11:25

4aSC11. Lexical representation of perceptually difficult second-language words. Mirjam Broersma (Max Planck Institute for Psycholinguistics, PO Box 310, Nijmegen 6500 AH, Netherlands, mirjam.broersma@mpi.nl)

This study investigates the lexical representation of second-language words that contain difficult to distinguish phonemes. Dutch and English listeners' perception of partially onset-overlapping word pairs like DAFFOdil-DEFIcit and minimal pairs like flash-flesh, was assessed with two cross-modal priming experiments, examining two stages of lexical processing: activation of intended and mismatching lexical representations (Exp.1) and competition between those lexical representations (Exp.2). Exp.1 shows that truncated primes like daffo- and defi- activated lexical representations of mismatching words (either deficit or daffodil) more for L2 than L1 listeners. Exp.2 shows that for minimal pairs, matching primes (prime: flash, target: FLASH) facilitated recognition of visual targets for L1 and L2 listeners alike, whereas mismatching primes (flesh, FLASH) inhibited recognition

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consistently for L1 listeners but only in a minority of cases for L2 listeners; in most cases, for them, primes facilitated recognition of both words equally strongly. Importantly, all listeners experienced a combination of facilitation and inhibition (and all items sometimes caused facilitation and sometimes inhibition). These results suggest that for all participants, some of the minimal pairs were represented with separate, native-like lexical representations, whereas other pairs were stored as homophones. The nature of the L2 lexical representations thus varied strongly even within listeners.

11:40–12:00 Panel Discussion

THURSDAY MORNING, 25 OCTOBER 2012 MARY LOU WILLIAMS A/B, 9:00 A.M. TO 11:45 A.M.

Session 4aSP

Signal Processing in Acoustics and Underwater Acoustics: Localizing, Tracking, and Classifying Acoustic Sources

Altan Turgut, Chair
Naval Research Lab, Washington, DC 20375

Contributed Papers

9:00

4aSP1. Passive sonar target tracking with a vertical hydrophone array in a deep ocean environment. Sheida Danesh and Henrik Schmidt (Massachusetts Institute of Technology, Cambridge, MA 02139, sdanesh@mit.edu)

When operating in a deep ocean environment, limited power availability makes it imperative to conserve energy. This is achieved through the use of computational efficiency, as well as a passive sonar configuration that eliminates the need for a sonar source. Mallat and Zhang's Matching Pursuits algorithm with a Kalman filter is implemented for use in passive target tracking. This makes it possible to determine the range of a moving target through the use of dot products and other simple calculations. The model setup used to test this approach includes a vertical hydrophone array at a depth of 4-5km and a near surface target between 10 and 45 km away. Simulated results using ray tracing (BELLHOP) and wavenumber integration (OASES) were used in developing this method. Preliminary results indicate this to be an effective means of target tracking. Possible future improvements include determining the bearing as well as the range of the target.

9:15

4aSP2. Autonomous underwater vehicle localization using the acoustic tracking system. Nicos Pelavas, Garry J. Heard, and Carmen E. Lucas (DRDC Atlantic, 9 Grove St., Dartmouth, NS B3A 3C5, Canada, nicos.pelavas@drdc-rddc.gc.ca)

Operator peace-of-mind during Autonomous Underwater Vehicle (AUV) missions is dependent on the ability to localize the vehicle. During launch and recovery phases this capability is particularly important. Defence R&D Canada (DRDC) Atlantic has designed and built a long-range tracking system for the International Submarine Engineering Explorer class AUVs. The acoustic tracking system (ATS) enables an operator on a loud icebreaker platform to determine the position of the AUVs at ranges up to 30 km. An acoustic projector, mounted on the AUV, emits a hyperbolic frequency modulated (HFM) chirp at a preset time interval. A small, directional, acoustic receiving array mounted near the stern of the icebreaker, accurately synchronized with the remote projector, receives signals from the distant AUV. Matched filter processing is used to determine the time of flight of the transmitted chirp. A beamforming algorithm applied to the data provides bearing and elevation angle estimates for the received signals. A ray tracing algorithm then uses this information, along with the sound velocity profile, to determine the position of the AUV. Moreover, ATS uses different HFM chirps to provide a basic one-way AUV state messaging capability. We conclude with a brief discussion of ATS data collected during in-water trials.

9:30

4aSP3. Passive localization of surface vessels in shallow water using broadband, unintentionally radiated noise. Alexander W. Sell and R. Lee Culver (Acoustics, Penn State University, State College, PA 16801, aws164@psu.edu)

The waveguide invariant relates ocean waveguide propagation conditions to the spectral interference patterns (or striations) in range-frequency plots. The striations are the result of interaction between propagating modes. A method of source localization, using a horizontal line array (HLA), that exploits this relationship will be presented. Source azimuth is estimated using conventional Bartlett beamforming, after which source range is estimated from spectral interference observed along the HLA as well as knowledge of the waveguide invariant. Automation of this process makes use of a spectral characterization method for striation slope estimation, which works well in some but not all cases. The use of a physics-based, range-dependent waveguide invariant model to improve the range estimates will also be discussed. This method has been applied to acoustical data recorded in 2007 at the Acoustical Observatory off the coast of Port of the Everglades, Florida. Localization results compare favorably with radar-based Automatic Identification System (AIS) records. [Work supported by ONR Undersea Signal Processing.]

9:45

4aSP4. Depth discrimination using waveguide invariance. Altan Turgut and Laurie T. Fialkowski (Naval Research Lab, Acoustics Div., Washington, DC 20375, altan.turgut@nrl.navy.mil)

Waveguide invariant theory is used to analyze the acoustic striation patterns generated by a moving surface vessel and a towed broadband (350-600 Hz) source during two field experiments (TAVEX08, AWIEX09) conducted in the East China Sea and New Jersey Shelf. Results from the East China Sea site indicated that slopes of striation patterns are different when the source is below the thermocline and receivers are below and above the thermocline. However, slopes are the same when the source (surface vessel) is above the thermocline and receivers are below and above the thermocline. In addition, results from the New Jersey Shelf site indicated that slopes of striation patterns are different when two co-located sources (tow-ship and towed source) are placed below and above the thermocline, and received on a single hydrophone below the thermocline. Results are explained by the dominance of reflecting and refracting modes for sources being above or below the thermocline during summer profile conditions. [Work supported by the Office of Naval Research.]

10:00

4aSP5. Application of a model-based depth discriminator to data from the REP11 experiment. Brett E. Bissinger and R. Lee Culver (Graduate Program in Acoustics, The Pennsylvania State University, PO Box 30, State College, PA 16804, beb194@psu.edu)

We address application of a passive, model-based depth discriminator to data from the REP11 experiment. The method is based on a mode subspace approach (Premus, 2007) which uses environmental information along with a normal mode based acoustic simulation to predict the propagating mode structure. This mode space can be divided into subspaces representing the lower and higher order modes. Sufficient aperture yields orthogonal and linearly independent subspaces and a linear algebraic process yields orthogonalized subspaces with reduced aperture. Received data is then projected onto these subspaces and a discrimination statistic is formed. This work examines the application of this process to data from the REP11 experiment in terms of performance of the discriminator over different sets of data and levels of environmental knowledge. Work sponsored by ONR Undersea Signal Processing.

10:15–10:30 Break

10:30

4aSP6. Sound speed estimation and source localization with particle filtering and a linearization approach. Tao Lin and Zoi-Heleni Michalopoulou (Department of Mathematical Sciences, New Jersey Institute of Technology, 323 ML King Blvd, Newark, NJ 07102, michalop@njit.edu)

In previous work, a particle filtering method was developed that provided estimates of multipath arrival times from short-range data and, subsequently, employed them in geometry, bathymetry, and sound speed inversion. The particle filter provided probability density functions of arrival times, that were then "propagated" backwards through a sound propagation model for inversion. That implies that every particle from the probability density is employed in the inversion scheme, creating a potentially computationally cumbersome process. In this work, we develop a new method for such parameter estimation which relies on linearization. The novel aspect is that the Jacobian matrix now includes derivatives with respect to Empirical Orthogonal Function coefficients. The approach, requiring only a few iterations to converge, is particularly efficient. Results from the application of this technique to synthetic and real (SW06) data are presented and compared to full-field inversion estimates. [Work supported by ONR and the NSF CSUMS program.]

10:45

4aSP7. Bayesian localization of acoustic sources with information-theoretic analysis of localization performance. Thomas J. Hayward (Naval Research Laboratory, 4555 Overlook Ave SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Approaches investigated to date for localizing acoustic sources include conventional beamforming, matched field processing, and Bayesian methods [e.g., Pitre and Davis, J. Acoust. Soc. Am., 97, 1995], with recent research revisiting Bayesian methods with focalization and marginalization approaches [Dosso and Wilmut, J. Acoust. Soc. Am., 129, 2011]. Information-theoretic bounds on source localization performance were investigated by Meng and Buck [IEEE Trans. Sig. Proc., 58, 2010] extending earlier work of Buck. The present work investigates direct application of Bayes' Rule to source localization and information-theoretic quantification and analysis of localization performance, taking as an example the localization of a time-harmonic source in a range-independent shallow-water acoustic waveguide. Signal propagation is represented by normal modes, and additive Gaussian ambient noise is represented by a Kuperman-Ingenito model. The localization performance is quantified by the entropy of the Bayesian posterior pdf of the source location, and an information-theoretic interpretation of this performance measure is presented. Comparisons with matched-field localization performance and extensions of the modeling and localization performance analysis to inhomogeneous media are discussed. [Work supported by ONR.]

11:00

4aSP8. Acoustic cavitation localization in reverberant environments. Samuel J. Anderson (The Graduate Program in Acoustics, The Pennsylvania State University, State College, PA 16801, sja183@psu.edu), Daniel A. Perlitz (Engineering Sciences, The Pennsylvania State University, State College, PA), William K. Bonness, and Dean E. Capone (Noise Control and Hydroacoustics, Applied Research Laboratory - PSU, State College, PA)

Cavitation detection and localization techniques generally require visual access to the fluid field, multiple high-speed cameras, and appropriate illumination to locate cavitation. This can be costly and is not always suitable for all test environments, particularly when the bubble diameter is small or duration is short. Acoustic detection and localization of cavitation can be more robust and more easily implemented, without requiring visual access to the site in question. This research utilizes the distinct acoustic signature of cavitation events to both detect and localize cavitation during experimental water tunnel testing. Using 22 hydrophones and the processing techniques plane-wave beamforming and Matched-Field Processing (MFP), cavitation is accurately and quickly localized during testing in a 12" diameter water tunnel. Cavitation is induced using a Nd:YAG laser for precise control of bubble location and repeatability. Accounting for and overcoming the effects of reflections on acoustic localization in acoustically small environments is paramount in water tunnels, and the techniques employed to minimize error will be discussed.

11:15

4aSP9. Doppler-based motion compensation algorithm for focusing the signature of a rotorcraft. Geoffrey H. Goldman (U.S. Army Research Laboratory, 2800 Powder Mill Road, Adelphi, MD 20783-1197, geoffrey.h.goldman.civ@mail.mil)

A computationally efficient algorithm was developed and tested to compensate for the effects of motion on the acoustic signature of a rotorcraft. For target signatures with large spectral peaks that vary slowly in amplitude and have near constant frequency, the time-varying Doppler shift can be tracked and then removed from the data. The algorithm can be used to preprocess data for classification, tracking, and nulling algorithms. The algorithm was tested on rotorcraft data. The average instantaneous frequency of the first harmonic of a rotorcraft was tracked with a fixed-lag smoother. Then, state space estimates of the frequency were used to calculate a time warping that removed the effect of the Doppler shift from the data. The algorithm was evaluated by analyzing the increase in the amplitude of the harmonics in the spectrum of a rotorcraft. The results depended upon the frequency of the harmonics, processing interval duration, target dynamics, and atmospheric conditions. Under good conditions, the results for the fundamental frequency of the target (~11 Hz) almost achieved the predicted upper bound. The results for higher frequency harmonics had larger increases in the amplitude of the peaks, but significantly fewer than the predicted upper bounds.

11:30

4aSP10. Automated entropy-based bird phrase segmentation on sparse representation classifier. Ni-Chun Wang, Lee Ngee Tan, Ralph E. Hudson (Electrical Engineering, University of California at Los Angeles, Westwood Plaza, Los Angeles, CA 90095, nichun@ucla.edu), George Kossan (Ecology and Evolutionary Biology, University of California at Los Angeles, Los Angeles, CA), Abeer Alwan, Kung Yao (Electrical Engineering, University of California at Los Angeles, Los Angeles, CA), and Charles E. Taylor (Ecology and Evolutionary Biology, University of California at Los Angeles, Los Angeles, CA)

An automated system capable of reliably segmenting and classifying bird phrases would help analyze field recordings. Here we describe a phrase segmentation method using entropy-based change-point detection. Spectrograms of bird calls are often very sparse while the background noise is relatively white. Therefore, considering the entropy of a sliding time- frequency window on the spectrogram, the entropy dips when detecting a signal and

rises back up when the signal ends. Rather than a simple threshold on the entropy to determine the beginning and end of a signal, a Bayesian recursion-based change-point detection(CPD) method is used to detect sudden changes in the entropy sequence. CPD reacts only to those statistical changes, so generates more accurate time labels and reduces the false alarm rate than conventional energy detection methods. The segmented phrases are then used for training and testing a sparse representation(SR) classifier, which performs phrase classification by a sparse linear combination of feature vectors in the training set. With only 7 training tokens for each phrase, the SR classifier achieved 84.17% accuracy on a database containing 852 phrases from Cassins Vireo (Vireo casinii) phrases that were hand-classified into 32 types. [This work was supported by NSF.]

THURSDAY MORNING, 25 OCTOBER 2012 BENNIE MOTEN A/B, 8:30 A.M. TO 11:30 A.M.

Session 4aUW

Underwater Acoustics and Acoustical Oceanography: Sources, Noise, Transducers, and Calibration

Ching-Sang Chiu, Chair
Department of Oceanography, Naval Postgraduate School, Monterey, CA 93943-5193

Contributed Papers

8:30

4aUW1. The measured 3-D primary acoustic field of a seismic airgun array. Arslan M. Tashmukhambetov, George E. Ioup, Juliette W. Ioup (Department of Physics, University of New Orleans, New Orleans, LA 70148, geioup@uno.edu), Natalia A. Sidorovskaia (Physics Department, University of Louisiana at Lafayette, Lafayette, LA), Joal J. Newcomb (Naval Oceanographic Office, Stennis Space Center, MS), James M. Stephens, Grayson H. Rayborn (Department of Physics and Astronomy, University of Southern Mississippi, Hattiesburg, MS), and Phil Summerfield (Geodetics & Cartography, ExxonMobil Corporation, Houston, TX)

The Littoral Acoustic Demonstration Center has conducted an experiment to measure the 3-D acoustic field of a seismic airgun array in the Gulf of Mexico. A seismic source vessel shot specified lines to give solid angle and range information. Hydrophone positions were measured by an ultra-short baseline (USBL) acoustic system while the source ship was turning between lines. An acoustic Doppler current profiler measured currents so the positions could be modeled between USBL measurements. The position locations were refined by using information from the acoustic arrival times on the hydrophones. Peak pressures, sound exposure levels, total shot energy spectra, one-third octave band analyses, and source directivity studies are used to characterize the field. One third octave band analysis shows received levels up to 180 dB re 1 μP for emission angles from 0 degrees (vertically down) up to 45 degrees for horizontal ranges up to 200 m at endfire, between 10 Hz and 200 Hz. The levels decrease with increasing frequency above 200 Hz, with increasing horizontal ranges, and for emission angles above 45 degrees. The levels are lower at broadside than at endfire. [Research supported by the Joint Industry Programme through the International Association of Oil and Gas Producers.]

8:45

4aUW2. Investigation of a tunable combustive sound source. Andrew R. McNeese, Thomas G. Muir (Applied Res. Labs., The University of Texas at Austin, 10000 Burnet Rd, Austin, TX 78757, mcneese@arlut.utexas.edu), and Preston S. Wilson (Mech. Eng. Dept. and Applied Res. Labs., The University of Texas at Austin, Austin, TX)

The Combustive Sound Source (CSS) is a versatile underwater sound source used in underwater acoustics experiments. The source is comprised of a submersible combustion chamber which is filled with a combustive gas mixture that is ignited via spark. Upon ignition, the combustive mixture is converted into high temperature combustion byproducts which expand and ultimately collapse back to smaller volume than before ignition. Acoustic pulses are radiated by the bubble activity. The CSS can be used as a source for calibration, TL measurements, and bottom characterizations, and when deployed on the bottom can create seismic interface waves. Current environmental regulations and varying experimental needs require a tunable source that allows users to easily alter the source level, bandwidth, and signal duration. Current efforts have focused on altering the bubble growth and collapse in attempt to tune the radiated signals to meet various needs. Scale models have been constructed and tested in in-house tank experiments. Discussion will focus on the results of the study along with future plans for development and modeling.

9:00

4aUW3. Mitigation of underwater piling noise by air filled balloons and PE-foam elements as hydro sound dampers. Karl-Heinz Elmer (Off-Noise-Solutions GmbH, Leinstr. 36, Neustadt a. Rbge. 31535, Germany, karl-heinz.elmer@t-online.de), Jörg Gattermann, Christian Kuhn, and Benedikt Bruns (Inst. Soil Mechanics and Found. Engineering, Techn. Universität Braunschweig, Braunschweig, Nds, Germany)

Founding of offshore wind turbines by pile driving induces considerable underwater sound emissions that are potentially harmful to marine life. In Germany, the Federal Maritime and Hydrographic Agency (BSH) has set a standard level of 160 dB (SEL) at a distance of 750 m from pile driving. Effective noise reducing methods are necessary to keep this standard level. The new method of hydro sound dampers (HSD) uses curtains of robust air filled elastic balloons showing high resonant effects, similar to air bubbles, but also balloons with additional dissipative effects from material damping and special dissipative PE-foam elements to reduce impact noise. The resonance frequency of the elements, the optimum damping rate for impact noise, the distribution and the effective frequency range can be fully controlled, if the HSD-elements are fixed to pile surrounding fishing nets. HSD-systems are independent of compressed air, not influenced by tide currents and easy adaptable to different applications. The theoretical background, numerical simulations, laboratory tests and offshore tests of HSD-systems result in noise mitigations between 17 dB to 35 dB (SEL). The work is supported by the German Federal Environmental Ministry (BMU).

9:15

4aUW4. Mitigation of underwater radiated noise from a vibrating work barge using a stand-off curtain of large tethered encapsulated bubbles. Kevin M. Lee, Mark S. Wochner (Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758, klee@arlut.utexas.edu), and Preston S. Wilson (Mechanical Engineering Department and Applied Research Laboratories, The University of Texas at Austin, Austin, TX)

A stand-off curtain of encapsulated bubbles with resonance frequencies of approximately 50 Hz was used to attenuate radiated noise from a work barge vibrated by onboard rotating machinery in a lake experiment. The purpose of this experiment was to provide a scale-model of how a noise reduction system of tethered encapsulated bubbles would be deployed to mitigate noise from a shallow water drilling ship. The work reported here is an extension of previous tests which used an array of encapsulated bubbles attached directly to the bottom of the work barge to reduce the radiated sound levels [J. Acoust. Soc. Am. **131**, 3506 (2012)]. The design of the new stand-off encapsulated bubble curtain is described, including the finite-element model that was developed to aide in the design. The deployment and acoustic testing of the curtain are also described. Results from the tests demonstrate that the system is both practical to deploy and is effective in reducing the underwater noise radiated into the lake from the work barge. [Work supported by Shell Global Solutions.]

9:30

4aUW5. Shipping source level estimation for ambient noise forecasting. Jeffrey S. Rogers, Steven L. Means, and Stephen C. Wales (Naval Research Lab, 4555 Overlook Ave SW, Washington, DC 20375, jeff.rogers@nrl.navy.mil)

The ability to accurately estimate shipping source levels from ambient noise data is an essential step towards creating a forecast model of the ocean soundscape. Source level estimates can be obtained by solving the system of linear equations, governed by the sonar equation, that relate source level to transmission loss (TL) and beamformer response. In this formulation, beamformer response is known and TL can be modeled from ship positions that are determined by a fusion of automatic identification system (AIS) reports and local radar data. Different levels of environmental realism will be taken into account for the TL model by considering two ocean bottom profiles. In particular, a layered sand-limestone bottom and karst sand-limestone bottom will be used in comparison for both 2D and NX2D TL runs. Source levels must be constrained to be positive and are thus solved for with a non-negative least squares (NNLS) algorithm. Estimation of source levels on data collected during the 2007 shallow water array performance (SWAP) experiment will be presented. Simulated ambient noise forecasts for the different sediment profiles will then be compared to real data from the SWAP experiment. [This work was supported by ONR.]

9:45

4aUW6. Prediction of noise levels on accelerometers buried in deep sediments. William Sanders and Leonard D. Bibee (Seafloor Sciences, Naval Research Laboratory, Stennis Space Center, MS 39529, wsanders@nrlssc.navy.mil)

The noise field below 100 Hz for three-axis accelerometers buried in sediments is due primarily to shipping, and to a lesser extent wind. Both are generated near the surface. Hence a buried sensor observes noise from an area of the sea surface around it extending theoretically across the entire ocean. However, practically more distant noise sources diminish (even though the area increases with the square of the distance) with range so as to limit the “listening area”. Sensors buried in sediments cut off horizontally propagating noise and hence are relatively more sensitive to locally generated noise. An elastic parabolic equation model is used to model the responses of three axis accelerometers buried in sediments within a complex geologic environment. The effect of shear waves in surrounding structures are shown to significantly affect the noise field. Noise from distant sources received by buried sensors is shown to be as much as 20 dB lower than that on sensors in the water column.

10:00–10:15 Break

10:15

4aUW7. Low-frequency ambient noise characteristics and budget in the South China Sea basin. Ching-Sang Chiu, Christopher W. Miller, and John E. Joseph (Department of Oceanography, Naval Postgraduate School, 833 Dyer Road, Room 328, Monterey, CA 93943-5193, chiu@nps.edu)

A sound record measured by a moored hydrophone in the South China Sea basin was analyzed. Sampled at a rate of 1.6 kHz and with a duty cycle of approximately 1-min-on and 14-min-off, the measured time series captures the spectral characteristics and variability of the ambient noise in the less-than-800-Hz band over an annual cycle. Using a combination of automated and manual screening methods, the dominant regular and transient noise sources were identified and categorized, which include shipping, wind waves, seismic air-gun surveys, shots/explosives and sonar. Intermittent self noise (squeaking sounds) that prevailed at times during the passage of the very large-amplitude internal waves was also identified. In addition to the noise budget, the variability in the daily and monthly means and variances of the measured noise spectrum and band levels were examined. In order to gain insights into the predictability of the ambient noise field in this marginal sea, the interpretation of the data was facilitated with temperature records measured with moored instruments, wind and precipitation time series from the US Naval Operational Global Atmospheric Prediction System (NOGAPS), and vessel motion simulation based on historical shipping density and lane structure. [Research sponsored by the Office of Naval Research.]

10:30

4aUW8. Using hydroacoutic stations as water column seismometers. Selda Yildiz (Marine Physical Laboratory, Scripps Institution of Oceanography/UCSD, 9500 Gilman Dr, La Jolla, CA 92093-0238, syildiz@ucsd.edu), Karim Sabra (School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA), W. A. Kuperman, and LeRoy M. Dorman (Marine Physical Laboratory, Scripps Institution of Oceanography/UCSD, La Jolla, CA)

The Comrehensive Nuclear-Test-Ban Treaty Organization (CTBTO) maintains hydrophones that have been used to study icebergs and T-wave propagation. These stations consist of three hydrophones at about the depth of sound channel in a horizontal triangle array with 2 km sides. We have used data from these stations in the few tenths of a Hertz and below regime to study if we can effectively use these stations as water column seismometers. Among the processing performed was methods to effectively transform the hydrophone configurations to vector sensors. An assortment of signal processing on hydrocoustic data from the December 26th 2004 Great Sumatra Earthquake has been compared to seismograph data of the same event indicating, that the hydrophone stations can indeed be used as surrogate seismometers.

10:45

4aUW9. Hydrophone calibration using ambient noise. Kristy Castillo Moore (Sensors and SONAR Systems, Naval Undersea Warfare Center Division Newport, 27744 Bugg Spring Rd, Okahumpka, FL 34762, kristy.moore@navy.mil) and Steven E. Crocker (Sensors and SONAR Systems, Naval Undersea Warfare Center Division Newport, Newport, RI)

Hydrophone calibration typically requires a good signal-to-noise (SNR) ratio in order to calculate the free-field voltage sensitivity (FFVS). However, the SNR requirements can limit the calibration of hydrophones with low sensitivity, particularly in the low frequency range. Calibration methods using ambient noise in lieu of a generated signal will be explored at the Underwater Sound Reference Division (USRD) Leesburg Facility in Okahumpka, FL. The USRD Leesburg Facility is at a natural spring in rural central Florida and is one of the Navy’s quietest open water facilities with no boating noise, limited biological noise, an isolated location, low reverberation and an isothermal water temperature profile below 5 meters. Comparison calibrations will be made with two similar hydrophones using the ambient noise in the natural spring and the results will be compared to calibrations made with the same hydrophones using a generated signal.

11:00

4aUW10. Transducer models for simulating detection processes for underwater mining. Kyounghun Been, Hongmin Ahn (Mechanical Engineering, POSTECH, Pohang-si, Gyeongbuk, Republic of Korea), Hunki Lee, Eunghwy Noh, Won-Suk Ohm (Mechanical Engineering, Yonsei University, Seoul, Republic of Korea), and Wonkyu Moon (Mechanical Engineering, POSTECH, Pohang Univ. of Science, Hyoja-dong, Nam-gu, Pohang, Gyeongbuk 790-784, Republic of Korea, wkmoon@postech.ac.kr)

Numerical simulations on propagating and scattering processes of sound waves in water and sediment may be useful for designing a detection system for underwater mining. Here a transducer model is developed for the numerical simulation to implement radiating and receiving processes of transducers into numerical calculations. Since the Rayleigh integral approach is adopted for acoustic radiation, the accurate velocity profiles over the radiating surfaces of a transducer array should be estimated considering the mechano-acoustic interactions including the dynamics of unit drivers and the acoustic radiation loadings on the radiation surfaces. We adopted the approach that the surface velocity is calculated using the transducer model with the acoustic loading while the loading effects are estimated via calculating the radiation

impedance of transducer array using Rayleigh integrals. The estimated velocity profile of the transducer surface is used for calculating the accurate sound fields generated by the transducer array. A similar approach will be adopted for estimating receiving characteristics. [The Authors gratefully acknowledge the support from UTRC(Unmanned technology Research Center) at KAIST(-Korea Advanced Institute of Science and Technology), originally funded by DAPA, ADD in the Republic of Korea.]

11:15

4aUW11. Acoustic insertion loss measurement using time reversal focusing. Jianlong Li and Zhiguang He (Department of Information Science and Electronic Engineering, Zhejiang University, Hongzhou, Zhejiang, China, JLLi@zju.edu.cn)

Accurate measurement of acoustic insertion loss has important applications in evaluating the performance of acoustic filtering material. In a typical procedure for insertion loss measurement, two pulses are recorded: one without and one with the specimen inserted between the transmitters and receivers. The amplitude spectra of the two pulses are then used to determine the insertion loss, which is a function of frequency. The measurement with low frequencies is quite difficult because of the reverberation interference, which is induced by the sides of vessel where the absorption materials cannot work well and fail to produce an acoustic free field environment. This presentation presents a method which uses time reversal (TR) focusing technique to measure the insertion loss of acoustic filtering materials. The experiment results in a waveguide water tank show that the approach can achieve high signal-to-reverberation ratio in the measurement. Besides, TR focusing provides high resolution at the place of the specimen which reduces the requirement of the specimen size. [Work supported by the National Natural Science Foundation of China under grant no 61171147.]

THURSDAY AFTERNOON, 25 OCTOBER 2012 — BASIE A1, 1:30 P.M. TO 6:00 P.M.

Session 4pAA

Architectural Acoustics, Noise, and Signal Processing in Acoustics: Alternative Approaches to Room Acoustic Analysis

Timothy E. Gulsrud, Cochair
Kirkegaard Associates, 954 Pearl St., Boulder, CO 80302

David S. Woolworth, Cochair
Oxford Acoustics, 356 CR102, Oxford, MS 38655

Invited Papers

1:30

4pAA1. Using spherical microphone array beamforming and Bayesian inference to evaluate room acoustics. Samuel Clapp, Jonathan Botts (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Greene Building, Troy, NY 12180, clapps@rpi.edu), Anne Guthrie, Ning Xiang (Arup Acoustics, New York, NY), and Jonas Braasch (Graduate Program in Architectural Acoustics, Rensselaer Polytechnic Institute, Troy, NY)

The most well-known acoustical parameters - including Reverberation Time, Early Decay Time, Clarity, and Lateral Fraction - are measured using data obtained from omnidirectional or figure-of-eight microphones, as specified in ISO 3382. Employing a multi-channel receiver in place of these conventional receivers can yield new spatial information about the acoustical qualities of rooms, such as the arrival directions of individual reflections and the spatial homogeneity. In this research, a spherical microphone array was used to measure the room impulse responses of a number of different concert and recital halls. The data was analyzed using spherical harmonic beamforming techniques together with Bayesian inference to determine both the number of simultaneous reflections along with their directions and magnitudes. The results were compared to geometrical acoustic simulations and used to differentiate between listener positions which exhibited similar values for the standard parameters.

1:50

4pAA2. Two home-brewed microphone assemblies for performing arts spaces. David Conant (McKay Conant Hoover Inc, 5655 Lindero Canyon Rd, Suite 325, Westlake Village, CA 91362, dconant@MCHinc.com)

Two decades ago, MCH pressed into service a binaural head (rather identical to one of its principals) comprised of a human skull, paraffin wax and anatomically-correct pinnae. This has been found useful in our concert hall tuning exercises. Separately, during tuning exercises, acousticians on our team reported possible percussion echoes during amplified events at Los Angeles' new 1700-seat Valley Performing Arts Center. Anticipating a deeper forensic exercise rapidly looming, a highly directional parabolic microphone system was cobbled from ad hoc parts to quickly confirm (or not) the reports, identify problem surfaces and potential solutions, if required. The curious-appearing, but effective devices are described and their use discussed.

2:10

4pAA3. Analysis of concert hall acoustics using time-frequency and time-spatial responses. Jukka Pätynen, Sakari Tervo, and Tapio Lokki (Department of Media Technology, Aalto University School of Science, Konemiehentie 2, Espoo FI02150, Finland, jukka.patynen@aalto.fi)

A set of objective parameters (ISO3382-1:2009) is widely used for describing acoustic conditions in performance spaces. With few exceptions, they are based on integrating sound energy within moderate time intervals. In practice, different acoustic conditions can yield similar values for objective measures. A presented method of analyzing concert hall acoustics with respect to the time-frequency features aims to overcome the deficiencies of the objective parameters by conveying considerably more information in an uncomplicated form. This is achieved by visualizing the contribution of short time frames in the impulse response to the cumulative sound energy as a function of frequency. Particularly the early part of the room impulse response, including the influence of the seat dip effect is efficiently visualized. The method is applied to acoustic measurements conducted at five corresponding positions in six concert halls. It is shown that in addition to communicating standard monaural objective parameters, the visualizations from the method are connected with several features regarding the subjective impression of the acoustics. The time-frequency analysis is further extended into utilizing a recent sound direction estimation technique. Resulting time-directional visualization enables the accurate analysis of early reflections and their contribution to spatial sound.

2:30

4pAA4. The importance and feasibility of "Reflectivity" as an index applicable for architectural acoustic design. Sooch SanSouci (Acoustic Design International LLC, Bryn Mawr, PA) and Felicia Doggett (Metropolitan Acoustics, LLC, 40 W. Evergreen Ave., Suite 108, Philadelphia, PA 19118, felicia@metropolitanacoustics.com)

A major part of room acoustics concerns sound control within a space. This traditionally involves reverberation control, music envelopment, optimization of speech comprehension and privacy, noise control, spatial enhancement, modal and reflection control for rooms used in recording, mixing, editing, mastering, and measuring sounds and hearing. In all of these examples, room acoustics is influenced by early reflections from the interior surfaces, objects and geometry. The impedance of a surface generally varies in relation to the incident angle of sound waves, therefore knowing the true reflectivity of surfaces would be complimentary to sound decay time and absorption coefficients. This presentation reviews current trends in the measurement of sound reflectivity as well as many of the challenges involved in developing an approved laboratory measurement methodology. Not only a potentially important tool for acousticians, architects and designers, but a formally adopted "Reflectivity Index" would aid product design, research, and education in room acoustics. Examples of comparative measurements of materials ranging from porous media to continuous or perforated surface assemblies are examined and how the range of results might be unified as a single metric.

2:50

4pAA5. STI measurements in real time in occupied venues. Wolfgang Ahnert and Stefan Feistel (Ahnert Feistel Media Group, Arkonastr. 45-49, Berlin D-13189, Germany, wahnert@ada-amc.eu)

To measure impulse responses in rooms and free fields is a daily Job for acousticians but mainly this is done in unoccupied cases. Results for the occupied case are derived by use of simulation software or by estimations based on experience. Using a newly developed multithread algorithm speech, music or any other signals from a microphone input and from a mixing console can be utilized to obtain impulse response data for further evaluation. In a soccer stadium in presence of more than 50.000 visitors the measurements have been done even as a function of the degree of occupancy. The method will be explained in detail and the needed conditions are described. All factors of influence are discussed to obtain results to derive impulses responses and to calculate the STI values in post processing.

3:10–3:25 Break

3:25

4pAA6. Non-traditional in-room measurement and non-measurement approaches to existing room evaluations. Dawn Schuette, Carl Giegold, and Molly Norris (Threshold Acoustics LLC, 53 W Jackson Blvd, Suite 815, Chicago, IL, dschuette@thresholdacoustics.com)

Threshold Acoustics has experimented with a number of non-traditional approaches to the evaluation of existing interior spaces in recent years. These methods have varied widely in response to the unique circumstances of each project. The challenges and advantages of each room must be approached with an open mind, in terms of both measured and qualitative evaluation, to reach a firm understanding of the acoustic character of a space. This paper will discuss studies that have utilized techniques as diverse as theatrical lighting and gels as a means of fine-tuning ceiling reflectors to working directly with musicians to determine how rooms as a whole or, in some cases individual room elements, respond to the frequency content and directionality of specific instruments. We will also discuss the method involved in a recent study of the interaction between a room and its reverberation chambers that was performed to gain a more complete understanding of the complex ways they influence one another.

3:45

4pAA7. Using musical instruments for narrow band impulse excitation of halls to aid in acoustical analysis. David S. Woolworth (Oxford Acoustics, 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

This paper suggests that narrow band specific information of a room's acoustic behavior can be revealed through the use of musical instruments as impulse sources that would be otherwise buried in response to a broadband impulse excitation or other sequence. An advantage to this is discovering acoustic anomalies before data analysis, as well as subjective judgments to room response and stage acoustics to add to standardized and more advanced technical methods. A violin, snare drum, and double bass are examined in terms of directionality and frequency content with and an example hall is analyzed.

4:05

4pAA8. Recent experience with low frequency room acoustics measurements. Timothy E. Gulsrud (Kirkegaard Associates, 954 Pearl Street, Boulder, CO 80302, tgulsrud@kirkegaard.com)

Despite the fact that both orchestral and popular music contain important low frequency (i.e., below 100Hz) sound energy, room acoustics measurements and parameters for concert halls do not typically consider the frequency range below the 125Hz octave band. This has resulted in inadequate objective descriptors of bass response and, in some cases, misguided acoustic designs to obtain good bass response. In this paper we present low frequency data measured in several concert halls around the world and discuss various methods for acquiring and analyzing the data, with the aim of encouraging further research in this area.

4:25

4pAA9. Evaluation of room-acoustic modal characteristics from single-point measurements using Bayesian analysis. Wesley Henderson, Ning Xiang, and Jonathan Botts (Architectural Acoustics, Rensselaer Polytechnic Institute, 110 8th Street, Greene Building, Troy, NY 12180, wesley.henderson11@gmail.com)

Room mode analysis is an important element of architectural acoustic design. In small rooms especially, well-separated low frequency room modes can cause unpleasant aural effects, such as undesirable resonances and flutter echoes. Traditional room mode analysis is generally done using the discrete Fourier transform. This work proposes a time-domain modal analysis method by which the amplitudes, frequencies, and damping constants of the room under test can be directly determined for low modal frequencies by using a Bayesian inference algorithm on a single room impulse response (RIR). The method's time-domain, model-based approach allows the number of modes present in the RIR, as well as the amplitudes, frequencies, and damping constants for each mode, to be determined using Bayesian model selection and parameter estimation, respectively. The method uses Skilling's nested sampling algorithm to infer parameter values. Results indicate that the method is especially useful in rooms with closely-spaced modes at low frequencies.

Contributed Papers

4:45

4pAA10. Methods of discovering problem room modes. Ben Bridgewater (University of Kansas, Lawrence, KS 60045, BenBridgewater@gmail.com)

Determining the best course of action to eliminate problem room modes in the Kansas Public Radio Live Performance studio required a way of measuring room modes and determining the problem dimensions. This case study focuses on how the room modes where measured and the determination of the problem modes in the KPR studio.

5:00

4pAA11. A comparison of source types and their impacts on acoustical metrics. Keely Siebein (University of Florida, P.O. Box 115702, Gainesville, FL 32611, ksiebein@siebeinacoustic.com)

The purpose of this study is to compare acoustical measurements made with different source types in a relatively reverberant room to determine if ISO 3382 monaural acoustic parameters such as Reverberation Time (RT), Early Decay Time (EDT), and Clarity Index (C80), yield different results for natural acoustic source stimuli. The source stimuli used in the study included Maximum Length Sequences (MLS), a running train of speech, a running piece of music and a balloon pop. The scientifically calibrated method is then compared to acoustical measurements obtained from natural acoustic sources, which include anechoic recordings of voice and music played through a directional speaker in the front of the room to simulate activities that would normally take place in the room, such as a person speaking and music being played during a worship service. This analysis is performed to determine if there are differences in acoustic room parameters using natural acoustic sources. This study essentially compares the effects of different source stimuli on measured acoustic parameters. It was found that different source signals and receiver locations significantly affect the acoustic metrics derived from the acoustical measurements due to variations in frequency, level and directionality.

5:15

4pAA12. Detecting the angle of arrival of discrete echoes by measuring polar energy time curves in a contemporary church. Ted Pyper and Ray Rayburn (K2 Audio, 4900 Pearl East Cir., Suite 201E, Boulder, CO 80301, ted@k2audio.com)

In order to deal with speech intelligibility concerns from the main sound system at a contemporary church, the authors executed measurements in the main assembly hall to detect late arriving reflections at various positions in the audience and on the stage. By implementing Polar Energy Time Curves (PETC) at each measurement location, discrete reflections were identified by time arrival and also, more critically, by angle of arrival. The advantage of calculating the PETCs on site during the measurement session was that the authors could physically pinpoint the sources of discrete echoes by using a laser positioned at the center of the microphone armature used to take the measurements. This gave the authors immediate feedback to diagnose reflections and help select additional measurement positions within the room. With this information, the authors were able to identify room surfaces that contributed significantly to the late arriving echoes and specify appropriate sound absorptive treatments for these surfaces.

5:30

4pAA13. Measurements of the just noticeable difference for reverberation time using a transformed up–down adaptive method. Adam Buck, Matthew G. Blevins, Lily M. Wang, and Zhao Peng (Durham School of Architectural Engineering and Construction, University of Nebraska-Lincoln, Omaha, NE 68182, atbuck@unomaha.edu)

This investigation sought to measure the just noticeable difference (JND) for reverberation time (T30) using a rigorous psychophysical method, namely the transformed up-down adaptive method. In ISO 3382-1:2009, the JND for reverberation metrics is taken to be 5%, based on work by Seraphim (1958); however, others have suggested that the actual JND is higher. In this project, sound samples with varying T30 were auralized from impulse responses simulated in a realistically modeled performance space using ODEON. The model's absorption coefficients were uniformly varied across all surfaces and frequencies to achieve the desired T30s. Three reference reverberation times were utilized (one, two, and three seconds), and eight T30 cases spaced at 4% intervals both above and below each of the three reference T30s were created. Auralizations using a 500 ms white noise burst were presented in a computer-based testing program running a three-interval one-up two-down forced choice method, presented in a sound booth over headphones with flat frequency response. The program randomly interleaved six staircase sequences, three of which ascended and three of which descended towards each reference T30. Results averaged across 30 participants will be presented. [Work supported by a UNL UCARE Grant and the ASA Robert W. Young Award.]

5:45

4pAA14. The role of acoustical characteristics for enhancement of acoustic comfort in high-speed train passenger cars. Hyung Suk Jang, Jooyoung Hong, and Jin Yong Jeon (Architectural Engineering, Hanyang University, Seoul, Seongdong-gu 133791, Republic of Korea, jyjeon@hanyang.ac.kr)

The room acoustic environments in high-speed trains have been investigated to identify the design elements for the passenger cars to improve acoustic comfort. Both room acoustical and psychoacoustical parameters affected by the absorption coefficients of interior finish materials were measured at the height of passengers' ears. Room acoustical simulation was constructed based on the measurements to investigate the effect of design elements influencing acoustic quality in the carriage. Through computer simulation of the models with changes in acoustical properties such as absorption/diffusion coefficients of the interior surfaces, the effect of interior design components were investigated and classified to improve the speech privacy.

THURSDAY AFTERNOON, 25 OCTOBER 2012 JULIA LEE A/B, 2:00 P.M. TO 3:15 P.M.

Session 4pABa

Animal Bioacoustics, Acoustical Oceanography, Structural Acoustics and Vibration, Underwater Acoustics, and ASA Committee on Standards: Underwater Noise from Pile Driving II

Mardi C. Hastings, Cochair
George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405

Martin Siderius, Cochair
ECE Dept., Portland State Univ., Portland, OR 97201

Contributed Papers

2:00

4pABa1. Results of a scaled physical model to simulate impact pile driving. Katherine F. Woolfe and Mardi C. Hastings (George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

To achieve a more complete understanding of the parameters involved in the structural acoustics of impact pile driving, a scaled physical model was developed and tested. While the design of the scaled model has been presented previously (Woolfe et al., JASA 130: 2558, 2011), this presentation focuses on analysis of wall velocity data and intensity data obtained from experimental evaluation of the model. The energy contained in a control volume surrounding the pile and the energy exchanged across the surface of the control volume were estimated from near field intensity measurements. The amount of energy transferred to the fluid from the cylindrical shell structure during impact and the amount of energy transferred to the structure from the fluid immediately following impact were determined. Results indicate that the highly damped pressure waveform as observed in the water column of the scaled physical model as well as in field data is due primarily to the transfer of energy from the surrounding water back into the structure. [Work supported by the Georgia Institute of Technology and the Oregon Department of Transportation through a subcontract from Portland State University.]

2:15

4pABa2. Modeling and visualization of the underwater sound field associated with underwater pile driving. Dara M. Farrell and Peter H. Dahl (Department of Mechanical Engineering and Applied Physics Laboratory, University of Washington, Seattle, WA 98105, daraf@uw.edu)

As communities seek to expand and upgrade marine and transportation infrastructure, underwater noise from pile driving associated with marine construction is a significant environmental regulatory challenge. This work explores results of different transmission loss models for a site in Puget Sound and the effect of improved understanding of modeling on the extents of zones of influence. It has been observed that most of the energy associated with impact pile driving is less than about 1000 Hz. Here, analysis of the spectral content of pile driving noise is undertaken to ascertain the optimal surrogate frequency to model the broadband nature of the noise. Included is a comparison of a normal mode model, which is motivated by work presented by Reinhall and Dahl [JASA 130, 1209 (2011)], with other methods. A GIS (Geographic Information System) tool, ArcMap, is used to map the sound level over the bathymetry, which has proved to be a useful way of visualizing the impact of the noise. [Work supported by Washington Sea Grant.]

2:30

4pABa3. A model for passive underwater noise suppression by bubble curtains surrounding point or line sources in shallow water. Todd A. Hay, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Applied Research Laboratories, The University of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, hayta@arlut.utexas.edu)

Underwater noise generated by pile driving, rotating machinery, or towers supporting offshore wind turbines may disturb marine life and inhibit detection of coastal activities via passive sonar and seismic sensors. Noise abatement techniques have therefore been proposed to limit the propagation of such noise into the far field, and many of these employ a curtain of freely-rising bubbles or tethered encapsulated bubbles to surround the towers [Lee *et al*., J. Acoust. Soc. Am. **131**, 3507(A) (2012)]. An analytic model, based on a Green's function approach, is presented for the passive noise suppression provided by a discrete number of bubbles surrounding submerged point sources or pulsating cylindrical towers above horizontally-stratified layers of sediment. The sediment layers are modeled as viscoelastic media and the Green's function is derived via angular spectrum decomposition [Hay *et al*., J. Acoust. Soc. Am. **129**, 2477(A), (2011)]. Simulations in which the bubbles are assumed to react independently to the incident field will be compared to those in which bubble-bubble interaction is taken into account. The effects of bubble size distributions and void fractions on noise suppression will be investigated for different source configurations. [This work was supported by the Department of Energy under Grant DE-EE0005380.]

2:45

4pABa4. Reduction of underwater sound from continuous and impulsive noise sources using tethered encapsulated bubbles. Kevin M. Lee, Andrew R. McNeese, Mark S. Wochner (Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758, klee@arlut.utexas.edu), and Preston S. Wilson (Mechanical Engineering Department and Applied Research Laboratories, The University of Texas at Austin, Austin, TX)

Arrays of encapsulated bubbles have been shown to be very effective in reducing underwater sound radiated from various sources [J. Acoust. Soc. Am. **131**, 3356 (2012); J. Acoust. Soc. Am. **131**, 3506 (2012)]. These arrays have been used to treat both sources of noise and to protect a receiving area from external noise. The system provides noise reduction using the combined effects of bubble resonance attenuation and acoustic impedance mismatching. Results are reviewed from experiments where tethered encapsulated bubble arrays were used to reduce underwater sound levels from both continuous and impulsive sound sources. In one set of experiments, the encapsulated bubble array attenuated continuous wave sound from a compact electromechanical source. In the other set of experiments, the continuous source was replaced by a combustive sound source [IEEE J. Oceanic Eng. **20**, 311–320 (1995)], which was intended to simulate real-world impulsive noise sources such as impact pile driving or airguns used in seismic surveys. For both the continuous and impulsive sources, the encapsulated bubbles provided as much as 45 dB of reduction in the 10 Hz to 600 Hz frequency band. [Work supported by Shell Global Solutions and ARL IR&D program.]

3:00

4pABa5. Dynamic response of a fish swim bladder to transient sound. Shima Shahab and Mardi C. Hastings (George W. Woodruff School of Mechanical Engineering, Georgia Institute of Technology, Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

High-level underwater sound exposures from impact pile driving activities and seismic air guns can cause physical damage to fishes. Damage to biological tissues depends on the strain rate induced by the initial rapid rise and fall times of the received sound pulse. So a time-domain analysis is needed to understand mechanisms of tissue damage resulting from exposure to sound from transient acoustic sources. To address this issue, the frequency domain mathematical model originally developed by Finneran and Hastings (JASA **108**: 1308-1321, 2000) was modified to predict the response of a fish swim bladder and surrounding tissues to transient signals. Each swim bladder chamber was modeled as an elastic prolate spheroidal shell filled with gas and connected to other parts of the anatomy. Results of three case studies are presented showing correlation between the strain rate of the swim bladder wall and tissue damage reported in five different species of fish.

THURSDAY AFTERNOON, 25 OCTOBER 2012 LESTER YOUNG A, 1:00 P.M. TO 5:50 P.M.

Session 4pABb

Animal Bioacoustics: Terrestrial Passive Acoustic Monitoring II

Ann E. Bowles, Chair
Hubbs-SeaWorld Research Institute, San Diego, CO 92109

Chair's Introduction—1:00

Invited Papers

1:05

4pABb1. Using acoustical monitoring to assess mule deer behavior in response to natural gas development and noise disturbance in the Piceance Basin, Colorado. Emma Lynch (Biology, Colorado State University, 1201 Oakridge Drive, Suite 100, Fort Collins, CO 80526, emma_lynch@nps.gov)

Passive recording units are valuable tools for assessing wildlife behavior, and can be used to address questions on multiple scales, from individual to landscape. We used this versatile technique to explore one of the most pressing wildlife management issues in the intermountain west: broad scale energy development. Much of this development occurs on critical wildlife habitat, and has been shown to alter the physical and acoustical properties of landscapes at a rapid rate. We designed and packaged an inexpensive, collar-mounted recording device for monitoring mule deer (Odocoileus hemionus) behavior with respect to natural gas development in Northwestern Colorado. This presentation will provide a summary of collar design, data analysis, and preliminary results.

1:25

4pABb2. Peeling the onion: What alarm calls can reveal about mammalian populations. Stacie Hooper (Evolution and Ecology, University of California at Davis, One Shields Avenue, Davis, CA 95616, cetacea@calweb.com), Brenda McCowan (Population Health and Reproduction, University of California at Davis, Davis, CA), Toni Lyn Morelli, and Christina Kastely (Environmental Science, University of California at Berkeley, Berkeley, CA)

While individually distinctive vocalizations have been used as a tool for the conservation and management of bird populations, few studies have investigated the potential of a bioacoustic tool for use with terrestrial mammals. Even relatively simple signals, such as alarm calls, have been shown to contain different types of information, and can even be individually distinctive in their structure. Similarities in vocal structure among individuals may also reflect close kin relationships, as suggested by previous work. We explored the feasibility of a bioacoustic tool for monitoring mammalian populations by testing for the presence of individual, age-class, and sex-related information within the alarm calls of Belding's ground squirrels from several geographically distinct populations. A neural network was used to successfully classify alarm calls to individual, age class and sex, demonstrating that this species' alarm calls contain information regarding at least three different caller characteristics. We also found that acoustic similarity, as measured by an index of acoustic distance, was significantly correlated with genetic relatedness between individuals. This work indicates that vocalizations have the potential to provide more information to wildlife managers about terrestrial mammal populations than just species presence, and may even provide insight about the level of inbreeding.

1:45

4pABb3. Response of nesting northern Goshawks to logging truck noise Kaibab National Forest, Arizona. Teryl Grubb (Rocky Mountain Research Station, U.S. Forest Service, 2500 S. Pine Knoll Dr., Flagstaff, AZ 86001, tgrubb@fs.fed.us), Larry Pater (Engineer Research Development Center, (Retired), Champaign, IL), Angela Gatto (Kaibab National Forest, U.S. Forest Service, Fredonia, AZ), and David Delaney (Engineer Research Development Center, Construction Engineering Research Laboratory, Champaign, IL)

We recorded 94 sound/response events at northern goshawk (Accipiter gentilis) nests 167, 143, and 78 m from the nearest road in Jun 2010: 60 experimentally controlled logging trucks, plus 30 passing light aircraft, 3 cars, and 1 all-terrain vehicle (ATV). Logging truck noise levels varied among nest sites and with distance from roads ($F = 36.753$, $P < 0.001$, $df = 59$). Aircraft noise levels for each day of testing ranged between 45.6-67.9 dB, and varied little among test sites, 60.1-65.6 dB ($F = 2.008$, $P = 0.154$, $df = 29$). Our test logging truck (61.9 dB adjusted CLEQ) was no louder than passing aircraft (62.3 dB adjusted CLEQ; t-test, $P = 0.191$), which goshawks generally ignored. The logging truck resulted in 27% no response and 73% alert response; passing aircraft resulted in 90% no response and only 10% alert response; and 3 cars and 1 ATV, combined, resulted in 50% each for no response and alert response ($\chi 2 = 82.365$, $P < 0.005$). Goshawk alert response rates were inversely proportional to nest distance from the nearest road ($\chi 2 = 29.861$, $P < 0.005$). Logging truck noise had no detrimental effects on nesting northern goshawks on the Kaibab Plateau, Arizona.

2:05

4pABb4. Use of acoustics to quantify and characterize bullet overshot into sensitive wildlife areas. David Delaney (U.S. Army, ERDC/CERL, 2902 Newmark Drive, Champaign, IL 61821, david.delaney@erdc.usace.army.mil) and Tim Marston (U.S. Army, Fort Benning, GA)

Live-fire training exercises on military installations are known to impact tree health and can potentially affect nesting/foraging habitat and behavior of terrestrial animals, though few studies have attempted to quantify and characterize bullet overshot from military training operations downrange into sensitive wildlife areas. There is concern about the potential impact that downrange military munitions might have on the federally endangered Red-cockaded Woodpecker and its foraging and nesting habitat. Various anecdotal methods have been used in an attempt to document bullet overshot, but none of these methods have been shown to be effective. The objective of this project is to demonstrate that acoustical techniques can accurately and effectively record and characterize live-fire bullet overshot into sensitive wildlife areas downrange of active military ranges. This research is part of a long-term study on Fort Benning, GA to investigate how munitions fire affects Red-cockaded Woodpecker nesting/foraging habitat and nesting behavior, while also investigating the effectiveness of earthen berms at stopping bullets from entering downrange areas. Preliminary results and field protocols will be presented and discussed.

2:25

4pABb5. Determinants of accuracy in a terrestrial microphone array. Alan H. Krakauer (Department of Evolution and Ecology, University of California at Davis, 2320 Storer Hall, One Shields Ave, Davis, CA 95616, ahkrakauer@ucdavis.edu), John Burt (Department of Psychology, University of Washington, Seattle, WA), Neil Willits (Department of Statistics, University of California at Davis, Davis, CA), and Gail L. Patricelli (Department of Evolution and Ecology, University of California at Davis, 2320 Storer Hall, One Shields Ave, Davis, CA 95616)

Acoustic sensor arrays can allow researchers to localize the position of vocalizing animals. During the course of research on a threatened bird species, the greater sage-grouse, we developed a 24-channel wired array to non-invasively monitor male courtship displays at traditional display grounds (i.e. leks). Here we describe a study in which we localized repeated playbacks of four local species while varying speaker position, the number and arrangement of microphones, and accuracy of speed of sound and sensor location estimates. As expected, localization accuracy was lowest when the speaker was outside the array and when using a linear microphone arrangements. We found no overall effect of species identity in spite of strong differences in time and frequency structure of the playbacks, although we did find significant interactions of species with other factors in our analysis. Simulated errors in speed-of-sound-in-air and estimation of sensor position revealed that while localization was most accurate when these errors were small, localization was still possible even with relatively large errors in these two factors. While we hope these results will help researchers to design effective sensor arrays, specific outcomes will depend on study-specific factors as well as the specific sound processing and localization algorithms employed.

2:45

4pABb6. Passive acoustic monitoring of bullfrog choruses: Spontaneous and evoked changes in group calling activity. Andrea M. Simmons (Cognitive, Linguistic & Psychological Sciences, Brown University, Box 1621, Providence, RI 02912, Andrea_Simmons@brown.edu), Jeffrey M. Knowles (Neuroscience, Brown University, Providence, RI), Eva Jacobs (Cognitive, Linguistic & Psychological Sciences, Brown University, Providence, RI), and James A. Simmons (Neuroscience, Brown University, Providence, RI)

We developed a multiple-microphone array method for recording temporal and spatial interactions of groups of vocalizing male bullfrogs, and for analyzing how this chorusing behavior is perturbed by playbacks of modified frog calls. Chorusing bullfrogs were recorded over 3 nights (90 min sessions) using an array of ten MEMS microphones distributed along a 20-m sector beside a natural pond. Vocal responses were digitized at 50 kHz using Measurement Computing A-to-D boards and customized software on a Lenovo Thinkpad. Individual frogs were located by time-difference-of-arrival measurements at the array. Baseline chorus activity was recorded for 10-20 min before and after playbacks. Playbacks consisted of digitized exemplars of two natural 5-croak advertisement calls, which were manipulated by adding or subtracting spectral components or by introducing masking noise. Baseline chorus activity featured both alternation of calls, mostly between far neighbors, and overlapping of calls, mostly by near neighbors. Bullfrog evoked vocal responses

were modified by playbacks of stimuli with altered spectral components, suggesting that the animals perceived these spectral modifications. The array technique indicated that responses of far neighbors were often more strongly impacted by playbacks than those of near neighbors.

3:05–3:20 Break

3:20

4pABb7. Environmental determinants of acoustic activity in Iberian anurans. Rafael Marquez, Diego Llusia (Dept de Biodiversidad y Biologia Evolutiva, Fonoteca Zoológica, Museo Nacional de Ciencias Naturales CSIC, Jose Gutierrez Abascal 2, Madrid 28006, Spain, rmarquez@mncn.csic.es), and Juan Francisco Beltran (Dept of Zoology, University of Seville, Sevilla, Spain)

We monitored acoustic activity of populations of anurans (genera Hyla and Alytes) in the Iberian Peninsula (Spain and Portugal) in localities at thermal extremes of their distribution. Logistic and linear regression models revealed that the major social and environmental determinants of calling behavior (chorus recruitment and chorus duration) were similar over most populations. Chorus recruitment was less dependent on environmental factors than chorus duration, which was also influenced by chorus size. Seasonal variation of night temperatures in populations of Hyla showed wide overall ranges (above 11 °C) and gradual increases of the nightly mean (3-12 °C), which was positively associated with the day number in the breeding season. Within days, temperatures were typically close to their daily maximum at sunset, the initiation of calling activity. We compared ranges of calling temperatures among species, populations, and seasons over three years. We showed that calling temperature changed when anuran populations were subjected to different thermal environments. Species had wide calling temperatures ranges across their distribution. Interannual comparisons showed that both terrestrial and aquatic breeding anurans were active during extremely hot breeding seasons. Lower thermal thresholds for the onset of calling were different between conspecific populations, suggesting that other factors are needed to trigger reproduction.

3:40

4pABb8. Passive acoustic monitoring of fish in shallow water estuaries. Mark W. Sprague (Dept. of Physics, East Carolina University, Mail Stop 563, Greenville, NC 27858, spraguem@ecu.edu), Cecilia S. Krahforst (Coastal Resources Management Doctoral Program, East Carolina University, Greenville, NC), and Joseph J. Luczkovich (Inst. for Coastal Science and Policy and Dept. of Biology, East Carolina University, Greenville, NC)

Passive acoustic monitoring is a useful tool for studying soniferous fishes in shallow water estuaries. We have used a variety of techniques for monitoring the acoustic environment in the coastal waters of North Carolina (USA) to study fishes in the Family Sciaenidae (drums and croakers), which produce sounds with frequencies below 1000 Hz. We will present data recorded with hydrophones deployed from a small boat, a hydrophone array towed behind a boat, and remote data loggers. We have used passive acoustic recordings to study the distributions (large- and small-scale) and seasonality of acoustically active courtship and spawning behavior, acoustic interactions between predators and prey, the effects of noise from tugs and small boats on fish sound production, and relationships between fish sound production and environmental parameters such as temperature and salinity. One limitation on shallow-water acoustic monitoring is the sound propagation cutoff frequency, which depends on the water depth. All frequency components below the cutoff frequency decay exponentially with propagation distance. This limit on shallow-water sound propagation must be considered when selecting locations for acoustic monitoring and comparing recordings made in waters of different depths. We will explore the implications on acoustic monitoring due to the cutoff frequency.

4:00

4pABb9. Reproductive success of Mexican spotted owls (*Strix occidentalis lucida*) in relation to common environmental noise - biotic, non-military aircraft, and weather-related. Ann E. Bowles (Bioacoustics Laboratory, Hubbs-SeaWorld Research Institute, 2595 Ingraham Street, San Diego, CA 92109, abowles@hswri.com), Samuel L. Denes (Graduate Program in Acoustics, Pennsylvania State University, State College, PA), Chris Hobbs, and Kenneth J. Plotkin (Wyle, Arlington, VA)

From 2000 to 2005, noise in Mexican spotted owl habitat in the Gila National Forest, NM, was monitored using an array of Larson-Davis (LD) sound level meters (SLMs). Thirty-nine SLMs were deployed across a 20 km x 24 km area, collecting 2-s time interval data mid-April to July, resulting in over 350,000 hr of data. Time-history profiles could be used to attribute many events to sources reliably when SNR exceeded the background by 5-10 dB. The events were categorized as biotic (insects and chorusing birds), thunder, regional commercial jet aircraft, and local air traffic (recreational and firefighting). Measured by the proportion of 2-s samples with $L_{Aeq} > 60$ dB, biotic sources and thunder were the most important. Regional commercial jet traffic was the most significant anthropogenic source, accounting for 2% of the total. Based on cumulative sound exposure, thunder was the greatest contributor. Regression techniques were used to relate owl reproductive success to noise metrics by source. Biotic noise was the only significant correlate, highly and positively related to owl reproductive success. The most reasonable interpretation was a strong relationship between biotic noise and owl prey base [Work supported by U.S. Air Force ACC/CEVP.]

4:20

4pABb10. Assessing the effects of sound on a forest-nesting seabird, the threatened marbled murrelet *(Brachyramphus marmoratus)*. Emily J. Teachout (Washington Fish and Wildlife Office, U.S. Fish and Wildlife Service, 510 Desmond Drive SE, Suite 102, Lacey, WA 98503, emily_teachout@fws.gov)

The marbled murrelet is a forest-nesting seabird that is federally listed under the Endangered Species Act. This species is unusual in that it flies up to 70 miles inland to nest in mature trees rather than on shorelines near its foraging habitat. The first nest was not discovered until 1974, and much remains to be learned about this difficult-to-study species, including its basic hearing sensitivities and response to auditory stimuli. In evaluating the effects of federal actions on this species, we must assess the potential effects of anthropogenic sound from a variety of sources, both at-sea and in the forested environment. Over the past ten years, we have developed an approach for analyzing the effect of anthropogenic sound by conducting literature reviews, convening expert panels, and drawing from

information on other species. We address both impulsive and continuous sounds from sources including impact pile driving, blasting, heavy equipment noise, and sonar. Interim thresholds for expecting injurious effects from some of these sources are in use, and refinements to our analysis of forest-management activities were recently applied to landscape-scale consultations. The bioacoustic research needs of this unique species continue to emerge as we apply these approaches in both the aquatic and terrestrial environments where murrelets occur.

4:40

4pABb11. The American National Standards Institute/Acoustical Society of America new (draft as 5 May 2012) standard method to define and measure the background sound in quiet areas. Paul Schomer (Schomer and Associates Inc., 2117 Robert Drive, Champaign, IL 61821, schomer@SchomerAndAssociates.com) and Kurt Fristrup (Natural Sounds and Night Skies Division, National Park Service, Ft. Collins, CO)

This draft standard is a joint work effort of S3/SC1, the animal bioacoustics committee, and S12, the noise committee. The draft standard includes 3 major, distinct components: (1) a method to measure the background using unattended instruments, and based on L-levels with L-90 as the default; (2) a definition for ANS-weighted sound pressure level, which is simply the A-weighted sound level after deleting all of the sound energy in the 2-kHz octave band or above; and (3), requirements for monitoring the sound in parks and wilderness area. The background measurement procedure is mainly for low-noise residential areas and relevant to the siting of such installations as wind farms and power plants. The ANS-weighting is applicable to both measurement of background and monitoring in parks. The requirements for monitoring in parks and wilderness areas ensure that measurements adequately capture the range of natural ambient conditions. The draft standard provides for two grades of measurement/monitoring: engineering or survey. In addition, this is the first standard to clearly and unambiguously define and establish the requirements for the measurement of percentile levels (exceedance values).

5:00–5:50 Panel Discussion

THURSDAY AFTERNOON, 25 OCTOBER 2012 TRIANON B, 2:00 P.M. TO 4:00 P.M.

Session 4pBA

Biomedical Acoustics: Therapeutic and Diagnostic Ultrasound

Robert McGough, Chair
Department of Electrical and Computer Engineering, Michigan State University, East Lansing, MI 48824

Contributed Papers

2:00

4pBA1. Effect of skull anatomy on intracranial acoustic fields for ultrasound-enhanced thrombolysis. Joseph J. Korfhagen (Neuroscience Graduate Program, University of Cincinnati, 231 Albert Sabin Way CVC 3948, Cincinnati, OH 45267-0586, joekorf3@gmail.com), Jason L. Raymond (Biomedical Engineering Program, College of Engineering and Applied Science, University of Cincinnati, Cincinnati, OH), Christy K. Holland (Internal Medicine, Division of Cardiovascular Diseases, University of Cincinnati, Cincinnati, OH), and George J. Shaw (Emergency Medicine, University of Cincinnati, Cincinnati, OH)

Transcranial ultrasound improves thrombolytic drug efficacy in ischemic stroke therapy. The goal of this study was to determine the ideal ultrasound parameters for obtaining peak rarefactional pressures exceeding the stable cavitation threshold at the left anterior clinoid process (lACP) of the skull. This location is near the origin of the middle cerebral artery, a common site for ischemic stroke. For 0.5, 1.1 and 2.0-MHz ultrasound transducers, pulse repetition frequencies (PRF) ranging from 5.4-8.0 kHz were studied at a 50% duty cycle. Attenuation and ultrasound beam distortion were measured from a cadaveric human skull. Each transducer was placed near the left temporal bone such that the unaberrated maximum acoustic pressure would be located at the lACP. A hydrophone measured the acoustic field around the lACP. Free-field measurements were taken in the same locations to determine attenuation and beam focus distortion. For 5 skulls, the average pressure attenuation at the lACP was 68 ± 19, 91 ± 5.1, and $94\pm4.7\%$ for 0.5, 1.1, and 2.0 MHz, respectively. The degree of displacement of the beam focus depended on the skull properties, but not the center frequency nor PRF. In conclusion, lower frequencies exhibited lower attenuation and improved penetration at the lACP. This work was supported by NIH-3P50-NS044283-06S1.

2:15

4pBA2. Fiber-optic probe hydrophone measurement of lithotripter shock waves under *in vitro* conditions that mimic the environment of the renal collecting system. Guangyan Li, James McAteer, James Williams (Department of Anatomy and Cell Biology, Indiana University School of Medicine, 635 Barnhill Dr., Indinapolis, IN 46202, gyli@iupui.edu), and Michael Bailey (Applied Physics Lab, University of Washington, Seattle, WA)

The fiber-optic probe hydrophone (FOPH) is the accepted standard for characterization of lithotripter shock waves, and measurements are typically conducted in water in the unconstrained free-field. To sort out potential factors that may affect *in vivo* measurements within the collecting system of the kidney, we assessed for the effect of contact between the fiber tip and tissue as might occur when working "blind" within the urinary tract, and contamination of the fluid medium (isotonic saline) by blood as often occurs during endourological procedures. Studies were performed using a Dornier Compact-S electromagnetic lithotripter. Contact of the optical fiber with *ex vivo* kidney tissue lowered pressure readings. The effect was greatest (~45% reduction) when the fiber was oriented normal to tissue, but pressures were also reduced when a length of fiber rested parallel against tissue (~5-10% reduction). Placing the fiber tip near, but not touching tissue, increased variability in peak negative pressure (P^-). Adding porcine blood to the medium (up to 10% V/V) had no effect on readings. These findings suggest that position/orientation of the FOPH relative to surrounding tissue is critical and

4p THU. PM

must be controlled, but that micro-hematuria will not be a confounding factor for *in vivo* measurements. (NIH-DK43881)

2:30

4pBA3. Speckle generation and analysis of speckle tracking performance in a multi-scatter pressure field. Ayse Kalkan-Savoy (Biomedical Engineering, UMass-Lowell, 1 University Ave, Lowell, MA 01854, ayse.k.savoy@gmail.com) and Charles Thompson (Electrical and Computer Engineering, UMass-Lowell, Lowell, MA)

Speckle tracking imaging is used as a method to estimate heart strain. An analysis of accuracy of speckle tracking and its potential to be utilized in quantification of myocardial stress through estimation of heart motion is examined. Multiple scattering effects are modeled using the Kirchoff integral formulation for the pressure field. The method of Pade approximants is used to accelerate convergence and to obtain temporal varying characteristics of the scattered field. Phantoms having varied acoustical contrast media and speckle density are used in this study. The effectiveness of inter-image frame of correlation methods for estimating speckle motion in high contrast media is considered. (NSF Grant 0841392)

2:45

4pBA4. Analytical and numerical approximations for the lossy on-axis impulse response of a circular piston. Robert McGough (Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824, mcgough@egr.msu.edu)

In biological tissues, the frequency dependence of attenuation and speed of sound for ultrasound propagation is described by the power law wave equation, which is a partial differential equation with fractional time derivatives. As demonstrated previously, the time domain Green's function for the power law wave equation combined with the Rayleigh-Sommerfeld integral is an effective reference for calculations of the lossy on-axis impulse response of a circular piston. Using the result obtained from this reference, two different approximations to the lossy on-axis impulse response are evaluated. The first approximation is an analytical expression that is proportional to the difference between two cumulative distribution functions for maximally skewed stable probability densities. The second approximation numerically convolves the lossless impulse response with a maximally skewed stable probability density function. The results show that both approximations achieve relatively small errors. Furthermore, the analytical approximation provides an excellent estimate for the arrival time of the lossy impulse response, whereas the departure time of the lossy impulse response is more difficult to characterize due to the heavy tail of the maximally skewed stable probability density function. Both approximations are rapidly calculated with the STABLE toolbox. [supported in part by NIH Grant R01 EB012079.]

3:00

4pBA5. The lossy farfield pressure impulse response for a rectangular piston. Robert McGough (Department of Electrical and Computer Engineering, Michigan State University, 2120 Engineering Building, East Lansing, MI 48824, mcgough@egr.msu.edu)

The impulse response of the velocity potential is useful for computing transient pressures in lossless media, especially for calculations in the nearfield region. Closed form expressions for the lossless impulse response of the velocity potential in the nearfield are available for circular and rectangular transducers and for several other geometries. A closed form lossless farfield expression is also available for rectangular transducers. Typically, when the effects of attenuation are introduced, the numerical calculation is performed in the frequency domain, and the time response is obtained with an inverse fast Fourier transform. To derive an equivalent analytical result directly in the time domain, all path lengths that appear in the denominator scaling term of the lossy diffraction integral are treated as constants, and a binomial expansion is applied to the path length that appears in the time delay term. The resulting analytical expression, which describes the lossy farfield pressure impulse response, is directly expressed in terms of maximally skewed stable probability density functions and cumulative distribution functions. Results are compared with the Rayleigh Sommerfeld integral, and excellent agreement is achieved in the farfield region. [supported in part by NIH Grant R01 EB012079.]

3:15

4pBA6. Histological analysis of biological tissues using high-frequency ultrasound. Kristina M. Sorensen (Department of Mathematics and Statistics, Utah State University, 698 E 700 N, Logan, UT 84321, Kristina.Sorensen@aggiemail.usu.edu), Timothy E. Doyle, Brett D. Borget, Monica Cervantes, J. A. Chappell, Bradley J. Curtis, Matthew A. Grover, Joseph E. Roring, Janeese E. Stiles, and Laurel A. Thompson (Department of Physics, Utah Valley University, Orem, UT)

High-frequency (20-80 MHz) ultrasonic measurements have the potential to detect cancer and other pathologies within breast tissues in real time, and thus may assist surgeons in obtaining negative or cancer free margins during lumpectomy. To study this approach, ultrasonic tests were performed on 34 lumpectomy margins and other breast tissue specimens from 17 patients to provide pulse-echo and through-transmission waveforms. Time-domain waveform analysis yielded ultrasonic attenuation, while fast Fourier transforms of the waveforms produced first- and second-order ultrasonic spectra. A multivariate analysis of the parameters derived from these data permitted differentiation of normal, adipose, benign, and malignant breast pathologies. The results provide a strong correlation between tissue microstructure and ultrasonic parameters relative to the morphology and stiffness of microscopic features such as ductules, lobules, and fibrous structures. Ultrasonic testing of bovine heart, liver, and kidney tissues supports this correlation, showing that tissues having stiff fiber-like or filled-duct structures, such as myocardium or ductal carcinomas, display greater peak densities in the ultrasonic spectra than tissues with soft, open duct-like structures, such as kidney tissue or normal breast glands. The sensitivity of high-frequency ultrasound to histopathology may assist in eliminating invasive re-excision for lumpectomy patients. [Work supported by NIH R21CA131798.]

3:30

4pBA7. The design and fabrication of a linear array for three-dimensional intravascular ultrasound. Erwin J. Alles, Gerrit J. van Dijk (Laboratory of Acoustical Wavefield Imaging, Delft University of Technology, Delft, Zuid-Holland, Netherlands), Antonius van der Steen (Biomedical Engineering, Thorax Centrum, Erasmus MC, Rotterdam, Zuid-Holland, Netherlands), Andries Gisolf, and Koen van Dongen (Laboratory of Acoustical Wavefield Imaging, Delft University of Technology, Lorentzweg 1, Room D212, Delft, Zuid-Holland, Netherlands K.W.A., vanDongen@TUDelft.nl)

Current intravascular ultrasound catheters generate high resolution cross-sectional images of arterial walls. However, the elevational resolution, in the direction of the catheter, is limited, introducing image distortion. To overcome this limitation, we designed and fabricated a linear array which can be rotated to image a three-dimensional volume at each pullback position. The array consists of eight rectangular piezo-electric elements of 350 μm by 100 μm operating at a center frequency of 21 MHz with a fractional bandwidth of 80 %, separated by a kerf of 100 μm. The array has been tested on both an ex vivo bovine artery and phantoms and, using the real aperture of the array, axially densely sampled images of the artery are obtained in every position. The array consistently yields significantly higher resolution in longitudinal images and more detail in radial images compared to a conventional catheter.

3:45

4pBA8. Parametric imaging of three-dimensional engineered tissue constructs using high-frequency ultrasound. Karla P. Mercado (Department of Biomedical Engineering, University of Rochester, Rochester, NY 14627, karlapatricia.mercado@gmail.com), María Helguera (Center for Imaging Sciences, Rochester Institute of Technology, Rochester, NY), Denise C. Hocking (Department of Pharmacology and Physiology, University of Rochester, Rochester, NY), and Diane Dalecki (Department of Biomedical Engineering, University of Rochester, Rochester, NY)

The goal of this study was to use high-frequency ultrasound to nondestructively characterize three-dimensional engineered tissues. We hypothesized that backscatter spectral parameters, such as the integrated backscatter coefficient (IBC), can be used to quantify differences in cell concentration in engineered tissues. We chose the IBC parameter since it estimates the backscattering efficiency of scatterers per unit volume. In this study, acoustic fields were generated using single-element, focused transducers (center frequencies of 30 and 40 MHz) operating over a frequency range of 13 to 47

MHz. Three-dimensional engineered tissue constructs were fabricated with mouse embryonic fibroblasts homogenously embedded within agarose. Constructs with cell concentrations ranging from $1x10^4$ to $1x10^6$ cells/mL were investigated. The IBC was computed from the backscatter spectra, and parametric images of spatial variations in the IBC were generated. Results showed that the IBC increased linearly with cell concentration. Further, we demonstrated that parametric images detected spatial variations in cell concentration within engineered tissue constructs. Thus, this technique can be used to quantify changes in cell concentration within engineered tissues and may be considered as an alternative to histology. Furthermore, because this technique is nondestructive, it can be employed for repeated monitoring of engineered tissues throughout the duration of fabrication.

THURSDAY AFTERNOON, 25 OCTOBER 2012 LIDO, 1:30 P.M. TO 3:15 P.M.

Session 4pEA

Engineering Acoustics: Electromechanical Considerations in Transducer Design

R. Daniel Costley, Cochair
Geotechnical and Structures Lab., U.S. Army Engineer R&D Center, Vicksburg, MS 39180

Robert M. Koch, Cochair
Chief Technology Office, Naval Undersea Warfare Center, Newport, RI 02841-1708

Contributed Papers

1:30

4pEA1. Analysis of electromechanical parameters of thick rings under radial, axial, and circumferential modes of polarization. Sairajan sarangapani (Department of Electrical Engineering, University of Massachusetts, Dartmouth, Fall River, MA 02747, ssairajan@yahoo.com), Corey L. Bachand, Boris Aronov (BTech Acoustics LLC, Fall River, MA), and David A. Brown (Department of Electrical Engineering, University of Massachusetts, Dartmouth, Fall River, MA)

Piezoceramic short hollow cylinders and annular disks operating in the extensional mode of vibration are typically used where the thickness is small compared to its lateral dimensions. For a thin ring, strain is along the circumferential direction and the mechanical system is considered as one dimensional. But it is not clear as to what extent a ring can be considered as thin. This study calculates the electromechanical parameters of thick rings under different modes of polarization using the energy method. An analytical formulation is presented and expressions for the internal energies and the electromechanical parameters are derived by analyzing the mechanical system of vibration under different polarization. The resonance frequencies, effective coupling coefficient and correction factors for the various electromechanical parameters under different modes of polarization as a function of thickness along the radial direction are presented.

1:45

4pEA2. Free in-plane vibrations of annular sector plates with elastic boundary supports. Xianjie Shi (College of Mechanical and Electrical Engineering, Harbin Engineering University, No. 145 Nantong Street, Nangang District, Harbin, Heilongjiang 150001, China, xianjshi@hotmail.com), Wen L. Li (Department of Mechanical Engineering, Wayne State University, Detroit, MI, United Kingdom), and Dongyan Shi (College of Mechanical and Electrical Engineering, Harbin Engineering University, Harbin, Heilongjiang, China)

In this investigation, a generalized Fourier series method is proposed for the in-plane vibration analysis of annular sector plates with elastic restraints along each of its edges. The in-plane displacement fields are universally expressed as a new form of trigonometric series expansions with a drastically improved convergence as compared with the conventional Fourier series. The expansion coefficients are considered as the generalized coordinates, and determined using the Rayleigh-Ritz technique. Several examples are presented to demonstrate the effectiveness and reliability of the current method for predicting the modal characteristics of annular sector plates with various cutout ratios and sector angles under different boundary conditions. It is also shown that annular and circle plates can be readily included as the special cases of the annular sector plates when the sector angle is set equal to 360°.

2:00

4pEA3. Comparison of the electromechanical properties of bars vibrating in flexure under transverse, longitudinal, and tangential polarization. Sairajan sarangapani (Department of Electrical Engineering, University of Massachusetts, Dartmouth, Fall River, MA 02747, ssairajan@yahoo.com), Boris Aronov (BTech Acoustics LLC, Fall River, MA), and David A. Brown (Department of Electrical Engineering, University of Massachusetts, Dartmouth, Fall River, MA)

The calculation of electromechanical properties of stripe-electroded bar vibrating in flexure becomes complicated as it involves nonuniform electric field distributions in the poling and operational mode and nonuniform mechanical strain distributions. This study is an extension of the previous work [J. Acoust. Soc. Am. 130, 2394 (2011)] and involves the calculation of the electromechanical parameters of stripe-electroded bars vibrating in flexure. The contributions due to the prominent longitudinal 33 mode, transverse 31 mode and the shear 15 mode are taken into account and the corresponding expressions for the internal energy (electrical energy, electromechanical energy and mechanical energy) are derived under the assumption that the piezoelement is fully polarized. Results of calculations are presented for effective coupling coefficient as a function of various distances between the electrodes and are compared with single-sided stripe-electroded bar design and the traditional bimorph designs using transverse and longitudinal piezoelectric effect.

2:15

4pEA4. Capacitive micromachined ultrasound Doppler velocity sensor using a nickel on glass process. Minchul Shin, Zhengxin Zhao (Mechanical Engineering, Medford, MA), Paul DeBitetto (Draper Labs, Cambridge, MA), and Robert D. White (Mechanical Engineering, Tufts University, 200 College Ave, Medford, MA 02155, r.white@tufts.edu)

The design, fabrication, modeling and characterization of a small (1 cm^2 transducer chip) acoustic Doppler velocity measurement system using a capacitive micromachined nickel on glass ultrasound transducer array technology is described. The acoustic measurement system operates in both transmit and receive mode. The device consists of 168 0.6 mm diameter nickel diaphragms, and operates at approximately 180 kHz. Computational

predictions suggest that in transmit mode the system will deliver an 11 degree -3dB beamwidth ultrasound. Characterization of the cMUT sensor with a variety of testing procedures including acoustic testing, Laser Doppler Vibrometry (LDV), beampattern test, reflection test, and velocity testing will be shown. LDV measurements demonstrate that the membrane displacement at center point is 0.1 nm/V^2 at 180 kHz. During beampattern testing, the measured response was 0.1 mV_{rms} at the main lobe with 90 kHz drive at 20 Vpp (frequency doubling causes the acoustics to be at 180 kHz). The maximum range of the sensor is 1.7 m. Finally, a velocity sled was constructed and used to demonstrate measureable Doppler shifts at velocities from 0.2 m/s to 0.8m/s. Doppler shifts are clearly seen as the velocity changes.

2:30

4pEA5. An electro-mechanical model of a carbon nanotube with application to spectrum sensing. Kavitha Chandra, Armand Chery, Fouad Attioui, Charles Thompson (University of Massachusetts Lowell, 1 University Ave, Lowell, MA 01854, kavitha_chandra@uml.edu),

The transduction of resonant mechanical vibrations induced in a single walled carbon nanotube (CNT) with application to frequency and signal detection is investigated in this work. The CNT, clamped between two metallic source and drain junctions and suspended in a trench above a gate electrode behaves as a conducting channel, controlled by appropriate gate and drain voltages. The Euler-Bernoulli model of an elastic beam is applied to model transverse vibrations of the CNT, considering the nonlinear stretching effect of the beam. The solution approach utilizes a Chebyshev-Galerkin spectral expansion and captures the influence of the fundamental and higher harmonic spatial modes when subject to time-harmonic and stochastic loads distributed across the beam. The transverse vibrations of the CNT generate a time-varying capacitance between the CNT and the gate that leads to non-equilibrium potential and charge induced on the tube. A self-consistent approach using a ballistic semi-classical transport model is applied iteratively to compute the charge and current variations at the drain terminal. Application of the system as a signal demodulator is analyzed by mixing the drain current with a matched high-frequency carrier signal that drives the gate terminal.

2:45

4pEA6. Highly efficient piezoelectric micromachined ultrasonic transducer for the application of parametric array in air. Yub Je, Wonkyu Moon (Mechanical Engineering, Pohang Universitiy of Science and Technology, San 31, Hyojadong Namgu, Pohang, Kyungbuk 790-784, Republic of Korea, wkmoon@postech.ac.kr), and Haksue Lee (Agency for Defense Development, Changwon, Kyungnam, Republic of Korea)

A highly efficient piezoelectric micromachined ultrasonic transducer was achieved for the application of parametric array in air. It is not easy to generate high-intensity sound required for nonlinear interaction in air due to huge impedance mismatch between the air and the transducer. A thin-film transducer such as micromachined ultrasonic transducer can achieve high mechanoacoustic efficiency by reducing the mechanical characteristic impedance of the radiating plate. By theoretical analysis, the power efficiency of the micromachined ultrasonic transducer and its maximum achievable value were considered. Also, based on the theoretical model, the design issues which reduce the power efficiency and radiation intensity were listed and discussed. The effects of the leakage current through the parasitic impedances, resistance of the electrode pad, and unit-to-unit variation of the MUT array on transducer efficiency were verified as the problems. By adapting proper design approaches, a highly efficient pMUT array was designed, fabricated, and tested as a source transducer for parametric array in air. The pMUT array may promote the practical uses of a parametric array source in air.

3:00

4pEA7. Acoustic filter design for precise measurement of parametric array in near-field region. Yonghwan Hwang, Yub. Je (Mechnical engineering, Postech, Po hang, Kyung sang buk do, Republic of Korea), Wonho Kim, Heesun Seo (ADD, Chang won, Kyung sang nam do, Republic of Korea), and Wonkyu Moon (Mechnical engineering, Postech, Hyo ja dong, Nam gu, Po hang, Kyung sang buk do KS010, Republic of Korea, wkmoon@postech.ac.kr)

A parametric array is a nonlinear conversion process that generates a narrow beam of low-frequency sound using a small aperture and that is used in active underwater SONAR and communication systems. A parametric array generates a highly directional difference frequency wave (DFW) by nonlinear interaction of bi-frequency primary waves at higher frequencies. However, it is difficult to measure the parametric array effect in the near distance of the transducer, because the high-pressure level of the primary waves may cause the receiving transducer to provide with pseudo-sound signals of difference frequency by its nonlinearity. The pseudo-sound signals make it difficult to measure the real DFW due to the parametric array process. In this study, we confirmed the existence of pseudo-sound and its effects on measurement of the near-distance DFW by the parametric array using numerical simulations based on the KZK equation. Also, a newly designed acoustic filter was proposed to eliminate pseudo-sound signals. This new acoustic filter uses the resonance of a thickness extensional mode. With the acoustic filter, low frequencies (DFW) were passed and high frequencies (primary frequencies) were effectively reduced. Using this acoustic filter, the precise characteristics of the difference frequencies could be measured. [Work supported by ADD(UD100002KD).]

Session 4pMU

Musical Acoustics and Animal Bioacoustics: Sound Production in Organs, Wind Instruments, Birds, and Animals—Special Session in Honor of Neville Fletcher

Thomas D. Rossing, Cochair
Stanford University, Los Altos Hills, CA 94022

Joe Wolfe, Cochair
University of New South Wales, Sydney, NSW 2052, Australia

Chair's Introduction—1:55

Invited Papers

2:00

4pMU1. Neville Fletcher: Scientist, teacher, scholar, author, musician, friend. Thomas D. Rossing (Music, Stanford University, Stanford, CA 94305, rossing@ccrma.stanford.edu)

Neville has distinguished himself in many ways. He has received many awards from scientific societies in Australia, America, Europe, and Asia. His hundreds of publications, including seven books, deal with physics, meteorology, biology, acoustics, and even poetry. Today we focus on his contributions to acoustics and especially to the Acoustical Society of America and the Australian Acoustical Society, where he is especially remembered for his work in musical acoustics.

2:20

4pMU2. Employing numerical techniques in the analysis and design of musical instruments. Katherine A. Legge and Joe Petrolito (Civil Engineering and Physical Sciences, La Trobe University, PO Box 199, Bendigo, VIC 3552, Australia, k.legge@latrobe.edu.au)

In the simplest of terms, a musical instrument consists of a source of oscillation coupled to a resonating body. The exception to this is an idiophone such as a triangle or gong, where the vibrating source is able to be its own radiator of sound. Whatever the configuration, the radiating structure is generally not a simple shape easily represented by a mathematical formulation, and analytical solutions to the governing equations of even a simplified model are often not obtainable. Working with Neville Fletcher in the 1980's a personal computer was employed to undertake a time-stepping routine through the equations of a simplified model of a kinked metal bar, to depict nonlinear coupling of its modes of vibration. Similar analysis of a gong modelled by a spherical shell with a kinked edge was well beyond the available computing power. In this paper we illustrate how the development of computers and numerical techniques over the intervening thirty years means that we are now able to describe and analyse complex shapes and are encroaching on the domain of the instrument maker through the use of numerical optimisation for instrument design.

2:40

4pMU3. The physics and sound design of flue organ pipes. Judit Angster (Acoustics, Fraunhofer IBP, Nobelstr 12, Stuttgart 70569, Germany, angster@ibp.fhg.de), András Miklós (Applied Acoustics, Steinbeis Transfer Center, Stuttgart, Baden Wuerttemberg, Germany), Péter Rucz, and Fülöp Augusztinovicz (Dept. of Telecommunications, Budapest University of Technology and Economics, Budapest, Budapest, Hungary)

Sound design methods for flue organ pipes can be developed for the practical application in organ building due to the high performance of modern computers and the advanced level of the scientific research on flue organ pipe acoustics. The research of Neville Fletcher and the team around him has contributed in a high extent to the scientific understanding of the behaviour of flue organ pipes. By extending this knowledge sound design methods and dimensioning software for different special flue organ pipes have been developed in the frames of European research projects. As examples the following topics will be mentioned: -the development of optimal scaling and a software for designing the depth and width of wooden organ pipes without changing the sound character; -the development of optimal scaling, design and software of chimney flutes by means of appropriate laboratory experiments and computer simulations.

3:00

4pMU4. Coupling of violin bridge modes to corpus modes analytical model. Graham Caldersmith (Caldersmith Luthiers, 12 Main Street, Comboyne, NSW 2429, Australia, luthier@westnet.com.au)

The transmission of string vibration forces to the violin belly by the bridge is modulated by two principal bridge resonances around 3KHz and 6 KHz, frequency bands critical to tone perception by the human hearing system. Much music acoustics research has dealt with these bridge modes (and those of the 'cello and bass) and their influence on the excitation of the corpus modes at the bridge feet. This analytical treatment of the string to corpus transmission by the bridge is necessarily complex, but reveals factors in the process which explain

the action of the different strings through the bridge and the levels of corpus mode excitation amplified by the bridge modes. The theoretical predictions are tested against experimental responses with normal bridges and bridges blocked to eliminate the two important modes.

3:20–3:30 Break

3:30

4pMU5. The didjeridu: Relating acoustical properties to players' reports of performance qualities. John Smith, Guillaume Rey, and Joe Wolfe (Physics, University of New South Wales, Sydney, NSW 2052, Australia, john.smith@unsw.edu.au)

Relating objective acoustical measurements of an instrument, without a player, to the qualities reported by players is often a difficult goal in music acoustics. The didjeridu offers advantages in such a study because it is inherently 'blind'—neither player nor researcher knows what is inside—and because there are wide variations in objective parameters. Here, seven experienced players reported several qualities and overall quality of 38 traditionally made didjeridus whose acoustic impedance spectra and overall geometry were measured. The rankings for 'overtones', 'vocals', 'resonance', 'loudness' and overall quality were all negatively correlated with the characteristic impedance of the instrument, defined as the geometric mean of the first impedance maximum and minimum. 'Speed' was correlated positively with the frequency of the lowest frequency impedance peak, near which the instrument plays. Assessments of geometrically simple PVC pipes yielded similar results. The overall ranking was highest for instruments with a low magnitude impedance, particularly in the 1-2 kHz range. This is the range in which players produce a strong formant in the radiated sound by varying vocal tract resonances with comparable ranges of impedance. This study and the researchers were inspired by the pioneering research in music acoustics by Neville Fletcher.

3:50

4pMU6. Bilateral coordination and the motor basis of female preference for sexual signals in canary song. Roderick A. Suthers (Medical Sciences, Indiana University, Jordan Hall, Bloomington, IN 47405, suthers@indiana.edu), Eric Vallet, and Michel Kreutzer (Laboratoire d'Ethologie et Cognition Comparees, Universite Paris Ouest, Nanterre, France)

The preference of female songbirds for particular traits in songs of courting males has received considerable attention, but the relationship of preferred traits to male quality is poorly understood. There is evidence that some aspects of birdsong are limited by physical or physiological constraints on vocal performance. Female domestic canaries (Serinus canaria) preferentially solicit copulation with males that sing special high repetition rate, wide-band, multi-note syllables, called 'sexy' or A-syllables. Syllables are separated by minibreaths but each note is produced by pulsatile expiration, allowing high repetition rates and long duration phrases. The wide bandwidth is achieved by including two notes produced sequentially on opposite sides of a syrinx, in which the left and right sides are specialized for low or high frequencies, respectively. The temporal offset between notes prevents cheating by unilaterally singing a note on the left side with a low fundamental frequency and prominent higher harmonics. The syringeal and respiratory motor patterns by which sexy syllables are produced, support the hypothesis that these syllables provide a sensitive vocal-auditory indicator of a male's performance limit for the rapid, precisely coordinated inter-hemispheric switching, which is essential for many sensory and motor processes involving specialized contributions from each cerebral hemisphere.

4:10

4pMU7. Acoustical aspects of the flute. William Strong (Physics & Astronomy, Brigham Young Univ., C126 ESC, Provo, UT 84602, strongw@byu.edu)

Neville Fletcher's wide ranging interests in excitation mechanisms and sound production in musical instruments and other sound sources have resulted in many related reports and publications. The presenter had the good fortune to work with Fletcher while holding a Senior Fulbright Fellowship at the University of New England. The joint research on acoustical aspects of the flute resulted in two papers: "Acoustical characterization of flute head joints" (Fletcher, Strong, and Silk, JASA 71, 1255-1260) and "Numerical calculation of flute impedances and standing waves" (Strong, Fletcher, and Silk, JASA 77, 2166-2172). These two papers will be reviewed in the presentation.

4:30

4pMU8. Free reeds as pressure-controlled valves. James P. Cottingham (Physics, Coe College, 1220 First Avenue, Cedar Rapids, IA 52402, jcotting@coe.edu)

The analysis of excitation mechanisms in pressure-controlled wind instruments by Neville Fletcher in a 1979 paper [Acustica 43, 63-72 (1979)] and later papers has been applied in understanding the operation of free reed instruments. This analysis enables calculation of impedance curves for reed generators and yields the well-known result that an inward striking reed coupled to a pipe operates at a frequency below the reed resonance and below but close to the frequency of an impedance maximum of the pipe, while an outward striking reed has operating frequency above both the reed resonance and the frequency of the pipe impedance maximum. This is useful, since free reeds can function as either inward striking or outward striking reeds, depending on details of the reed design and possible coupling with resonators. Fletcher's analysis was first applied to free reeds in the harmonica by Johnston [Acoustics Australia, 16, 69-75 (1987)] and has subsequently been applied to free reeds and free reed instruments in a variety of ways, which are summarized in this paper. These include modeling the change in sounding frequency with blowing pressure as well as determining the sounding frequency of both inward and outward striking reed-resonator combinations.

4:50

4pMU9. K-12 acoustics experiments. Uwe J. Hansen (Indiana State University, Terre Haute, IN 47803, uwe.hansen@indstate.edu)

Neville Fletcher has been active in many areas of acoustics, including acoustics education. He chaired a committee charged with revamping the science education curriculum in Australia's elementary and secondary education programs. Neville believes, as do I, in the old maxim: "You hear - you forget, You see - you remember, You do - you understand". In keeping with that I will discuss a number of basic acoustics experiments adaptable in the elementary and secondary classroom or hands-on laboratory, including speed of sound,

musical intervals, the Doppler effect, beats, resonance, spectral analysis and synthesis, musical instruments (violin, clarinet, etc.), the human voice, and sound level.

5:10

4pMU10. The evolution of musical instruments. Neville H. Fletcher (Research School of Physics and Engineering, Australian National University, Canberra, ACT 0200, Australia, neville.fletcher@anu.edu.au)

The first musical instruments probably arose by chance because of the observed acoustic properties of materials, tools and weapons. Subsequent human generations have then refined their design to produce the broad array of instruments we know today. Percussion instruments based upon simple wooden slats evolved into marimbas, metal shields evolved into gongs and bells, the taut string of a military bow became a guitar and so on, while in the wind-instrument domain the uniform tubes of bamboo or animal leg bones became flutes, organs, or reed instruments. Hydraulic mechanisms, then electrical motors, developed to power instruments such as pipe organs, while the feedback oscillations of electric amplifiers gave rise to electronic instruments. It is not possible to cover all aspects of this evolution in a short presentation, but some interesting examples, particularly for wind instruments, will be examined in more detail. It will be seen that this evolution is not a terminated process but is still continuing today under influences very much like those of Darwinian natural selection.

5:30–5:45 Panel Discussion

THURSDAY AFTERNOON, 25 OCTOBER 2012 TRIANON C/D, 2:00 P.M. TO 5:05 P.M.

Session 4pNS

Noise, Architectural Acoustics, and ASA Committee on Standards: Ongoing Developments in Classroom Acoustics—Theory and Practice in 2012, and Field Reports of Efforts to Implement Good Classroom Acoustics II

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

Louis C. Sutherland, Chair
lcs-acoustics, 5701 Crestridge Ridge, Rancho Palos Verdes, CA 90275

Chair's Introduction—2:00

Invited Papers

2:05

4pNS1. Renovation of an elementary school to meet high performance acoustic prerequisites for a special high school: Design predictions and commissioning results. Steve Pettyjohn (The Acoustics & Vibration Group, Inc., 5700 Broadway, Sacramento, CA 95820, spettyjohn@acousticsandvibration.com)

The renovation of an elementary school to house the MET high school program was done with a goal of meeting High Performance/Green Building (HP/GB) requirements. Two acoustic prerequisites are a part of the HP requirements. If these two requirements are not met, the project can not meet HP criteria no matter the number of points achieved in other areas. Acoustical consulting was brought in at the 90 percent construction document phase with most designs completed for the mechanical system design, wall design and room finishes. An evaluation of the mechanical system, reverberation time and sound transmission loss showed that modifications would be required. The HVAC systems had been ordered and placement was fixed. A room mock-up was done in the school to show that mechanical noise would exceed the 45 dB(A) limit as designed and installed. Early morning tests done on a modified system showed that meeting the prerequisites for background sound would be very difficult without additional modifications. Wall, door and window modifications were implemented as was some of the acoustical treatment. Final sound commissioning tests proved that all of the prerequisites had been met and the low background sound from all sources was a pleasant surprise.

2:25

4pNS2. A soundscape study in an open plan classroom in a primary school. Sang Bong Shin and Gary W. Siebein (School of Architecture, University of Florida, PO Box 115702, Gainesville, FL 32611-5702, archisangbong@gmail.com)

Advantages and disadvantages on open plan classrooms have been reported since this type of classroom was first constructed in 1970's all over the world. A new type of open plan classroom combined with small classrooms that accommodate specific learning activities has been developed as the 21st century learning environment to foster creative, interactive learning. In this study, a new school

building which has this type of open plan classroom is examined with soundscape approaches in order to investigate the acoustical environments in the building. Acoustical events occurring in the open plan classroom in a primary school were analyzed and categorized. The activities that created specific acoustical events were observed and categorized and then acoustical taxonomies were established for each space. Acoustical measurements were conducted to examine the acoustical characteristics of each acoustic event and the combinations of acoustical events in the learning spaces. In addition, the results of the observations and measurements were compared to determine the differences in the effects of room conditions on the acoustical events in both traditional and open plan classrooms in the same school. The study found that there are differences in the characteristics of the acoustical events as measured in traditional and open plan according classrooms.

2:45

4pNS3. Room acoustic effects on speech comprehension by English-as-a-second-language versus native English-speaking listeners. Zhao Peng, Lily M. Wang, and Siu-Kit K. Lau (Durham School of Architectural Engineering and Construction, University of Nebraska-Lincoln, 1110 S. 67th Street, Omaha, NE 68182-0816, zpeng@unomaha.edu)

English-as-a-second-language (ESL) listeners are generally more impaired than native English-speaking listeners on speech intelligibility tasks under reverberation and noise; however, more study is needed to ascertain these groups' performances on speech comprehension tasks, rather than intelligibility. A recent study (Valente et al, 2012) showed that speech comprehension by both native English-speaking adults and children was more negatively affected by adverse acoustic environments than sentence recognition. The current project investigates the speech comprehension of both ESL and native English-speaking adult listeners under combinations of reverberation and noise. Sets of 15-minute long listening comprehension tests based on the format of the Test of English for International Communication (TOEIC) were developed, anechoically recorded, convolved with binaural room impulse responses (BRIRs) to produce five mid-frequency reverberation times (0.4 to 1.1 seconds), and presented over loudspeakers in the Nebraska acoustics chamber. Background noise was also varied at three levels (RC-30, 40 and 50) through an Armstrong i-Ceiling system. In total then, subjects were individually exposed to 15 acoustic combinations of reverberation and background noise. Preliminary results will be presented and compared between ESL and native English listeners. [Work supported by a UNL Durham School Seed Grant and the Paul S. Veneklasen Research Foundation.]

3:05–3:20 Break

3:20

4pNS4. The link between cooling mechanical system type and student achievement in elementary schools. Ana M. Jaramillo and Michael Ermann (School of Architecture + Design, Virginia Tech, Blacksburg, VA 24060, anaja@vt.edu)

In aggregate, air-conditioning systems are the largest contributors to classroom noise, and some types of air-conditioning systems are noisier than other types. The impact of noise in human performance has been widely studied and found to be stronger in children. A study of 73 elementary schools in a single Orlando, Florida school district sought to relate mechanical cooling system type with student achievement. It found that for schools populated with students of similar socio-economic background, schools cooling with the noisiest types of mechanical system underperformed on student achievement tests relative to those with quieter types of systems. It also found that schools with the poorest students were more likely to cool their classrooms with noisier systems.

3:40

4pNS5. Acoustic contributions and characteristics of floor treatments for elementary school classroom background noise levels. Robert Celmer, Clothilde Giacomoni, Alex Hornecker, Ari M. Lesser, Adam P. Wells, and Michelle C. Vigeant (Acoustics Prog. & Lab, University of Hartford, 200 Bloomfield Avenue, West Hartford, CT 06117, celmer@hartford.edu)

A two part investigation involving the effect of floor treatments on classroom background noise levels will be presented. Phase 1 determined the effects of hard versus soft flooring on overall speech and activity noise levels using long-term calibrated sound recordings in elementary classrooms. Two similar-sized classrooms were used: one with vinyl composition tile (VCT) flooring, and one with short-pile commercial carpeting. After parsing the recordings into separate segments of (a) teacher/student speech (alone), and (b) classroom activity noise, including footfalls, chair scrapes, and impacts (no speech), a significant decrease in overall levels was found in the carpeted rooms. Phase 2 determined the acoustical properties of nine different flooring materials ranging from resilient athletic floors to VCT to commercial carpeting. Sound absorption was measured following ISO 10534-2, while ISO 3741 sound power measurements were made while either (a) using a standard tapping machine, or (b) scraping a classroom chair back/forth over the floor surface in a reciprocating manner. In general, both carpet samples resulted in the lowest sound levels and the highest absorption. Relative performances of each material will be presented along with additional classroom usability factors, such as maintenance, cost and durability. [Work supported by Paul S. Veneklasen Research Foundation.]

Contributed Paper

4:00

4pNS6. Comparing performance of classrooms with similar reverberation times but varying absorptive material configurations. James R. Cottrell and Lily M. Wang (Durham School of Architectural Engr. and Constr., Univ. of Nebraska - Lincoln, Lincoln, NE 68182, Jamesrcottrell@gmail.com)

The ASA/ANSI Standard on Classroom Acoustics S12.60-2010 suggests that the mid-frequency reverberation times of classrooms be less than 0.6 seconds; however, no details are provided on how to place absorptive materials optimally within the classroom. In this investigation, impulse responses from a database measured at Armstrong World Industries have been analyzed to determine the effect of varying absorptive material configurations on resulting acoustic parameters, including reverberation time, clarity index (C50), and speech transmission index (STI). Two different material configurations were analyzed for each of three mid-frequency-averaged reverberation times: 0.6 sec, 0.8 sec, and 0.9 sec. Results show that metrics significantly differ when the material placement varies dramatically, and that configurations with more evenly distributed absorption produce better conditions for speech intelligibility throughout the classroom. [Work supported by a UNL Undergraduate Creative Activities and Research Experience Grant.]

4:15–5:05 Panel Discussion

THURSDAY AFTERNOON, 25 OCTOBER 2012 TRIANON A, 2:00 P.M. TO 4:00 P.M.

Session 4pPA

Physical Acoustics and Noise: Infrasound II

Roger M. Waxler, Chair
NCPA, University of Mississippi, University, MS 38677

Contributed Papers

2:00

4pPA1. Infrasound propagation in the atmosphere in a presence of a fine-scale wind velocity structure. Igor Chunchuzov, Sergey Kulichkov, Oleg Popov, Vitaly Perepelkin (Obukhov Institute of Atmospheric Physics, 3 Pyzhevskii Per., Moscow 119017, Russian Federation, igor.chunchuzov@gmail.com), Roger Waxler, and Jelle Assink (National Center for Physical Acoustics, University, MS)

The results of modeling infrasound propagation in the atmosphere from 100-t surface explosion in Israel, and volcano eruptions in Ecuador and Kamchatka are presented. The signal as a function of a range from a source is calculated by parabolic equation method for the vertical profile of wind velocity and temperature obtained from the Ground-to-Space atmospheric model. The effects of fine-scale layered structure of wind velocity and temperature fields on infrasound propagation in the atmosphere have been also taken into account. The one-dimensional (vertical and horizontal) wavenumber spectra of the wind and temperature fluctuations associated with the fine-scale structure are close to the observed spectra in the middle and upper atmosphere. The calculated wave forms, amplitudes and durations of the stratospheric and thermospheric arrivals are compared with those observed in the experiments. It is shown that the scattering of infrasonic signals by anisotropic fluctuations leads to a significant increase in the duration of the stratospheric and thermospheric arrivals in comparison with the case when these fluctuations are absent. The acoustic field scattered from anisotropic nonhomogenerities may be responsible for the arrivals of the acoustic pulses observed in the acoustic shadow zones.

2:15

4pPA2. An infrasound calibration system for characterizing the environmental effect on sensor sensitivity and noise floor. Carrick L. Talmadge (NCPA, University of Mississippi, 1 Coliseum Drive, University, MS 38655, clt@olemiss.edu)

An infrasound calibration system has been developed at the National Center for Physical Acoustics. The calibration tank is comprised of a 1" cylindrical shell 40" in diameter, 40" long, with 40" diameter hemispherical end caps. The interior volume of the tank is approximately 1.8 cubic meters. Each hemisphere has a 10" punch-out with sealing gasket that allows either a speaker assembly or an "end cap" to be attached to each end. Normal assess is through an end cap attached to one end, with a speaker assembly attached to the other. The end cap allows rapid switching out of sensors. The speaker assembly consists of a 10" subwoofer with a sealable back volume designed to equalize the static pressure in the interior of the tank to that of the speaker back volume. The subwoofer is able to generate pressures up to 10-Pa in the interior of the chamber, with excellent isolation from external sound. Using 2 subwoofers, by playing a different frequency tone through each speaker and measuring their associated intermodulation distortion in the transduced signal, the linearity of the sensor can be accurately assessed. The measured leak constant of the tank is longer than one week, permitting the characterization the sensor response to ambient pressure to be measured. The large interior volume of the calibration chamber allows the interior be heated or cooled returned slowly to ambient conditions, so that the effects of changing temperature on the sensor can be assessed.

2:30

4pPA3. Detection of infrasonic energy from tornado-producing storms. Carrick L. Talmadge (NCPA, University of Mississippi, 1 Coliseum Drive, University, MS 38655, clt@olemiss.edu), William G. Fraizer, Roger Waxler (NCPA, University of Mississippi, University, MS), Joseph C. Park (Center for Operational Oceanographic Products and Services, National Oceanic and Atmospheric Administration, Silver Spring, MD), Daniel E. Kleinert (NCPA, University of Mississippi, University, MS), Geoffery E. Carter, Gerald Godbold, David Harris, Chad Williams (Hyperion Technology Group, Inc, Tupelo, MS), and Hank R. Buchanan (NCPA, University of Mississippi, University, MS)

There are numerous reports in the literature on the observation of infrasound emitted from tornadic thunderstorms. Most of these observations have been made from sensors that are several hundreds of kilometers from the location of the storm, and "ground truth" about the tornadic activity is not well established. We report here on a campaign carried out during the

summer of 2011 in which 50 infrasound microphones were deployed, as part of an ongoing multi-university program on hazard detection and alert funded by the National Oceanic and Atmospheric Administration. Sensors were placed along the paths of developing tornadic storms. We focus here on a severe weather outbreak that took place near Oklahoma City on May 24, 2011, in which a total of 7 tornados including one F5 and two F2 tornados were produced. Three sensors were located between the paths of an F4 and an F5 tornado, and 11 additional sensors were located northeast of an F4 tornado that generated a 75-km track. Substantial meteorological information, including ground truth about tornados (intensity and size as a function of time), and the relative close proximity of the sensors to the storms, provides us with a level of detail not available in previous storms. We will report on our infrasound measurements and analysis from this outbreak as well as discuss data from two other interceptions of tornadic storms, which occurred on May 30 and June 19, 2011.

2:45

4pPA4. Geomagnetic auroral infrasound wave characteristics and generation. Justin J. Oldham, Charles R. Wilson, John V. Olson, Hans Nielsen, and Curt Szuberla (Physics, University of Alaska Fairbanks Geophysical Institute, PO Box 750972, Fairbanks, AK 99775, joldham6@alaska.edu)

Periods of persistent, high-trace velocity infrasound activity have been routinely observed in the data from the CTBT/IMS I53US infrasound station in Fairbanks, AK. Previous studies of magnetic field disturbances and displays of aurora borealis suggested that these infrasound signals were generated as bow waves by moving auroral electrojets. Recent analysis of the data obtained from the Geophysical Institute Magnetometer Array, the Poker Digital All-Sky Camera, and historic data from the Poker Flat Imaging Riometer have demonstrated the presence of extended periods of infrasound signals during times of enhanced geomagnetic activity along with verification that the observed infrasound is being generated in the lower ionosphere. Further examination of these data sets and the I53US infrasound data provide a basis for comparison with idealized magneto-hydrodynamic models of geomagnetic auroral infrasound wave generation.

3:00

4pPA5. Non-linear infrasound signal distortion and yield estimation from stratospheric and thermospheric arrivals. Joel B. Lonzaga, Roger Waxler, and Jelle Assink (National Center for Physical Acoustics, University of Mississippi, 1 Coliseum Dr., University, MS 38677, jblonzag@olemiss.edu)

The propagation of sound through gases and fluids is intrinsically non-linear. The degree of non-linearity increases as the density of the propagation medium decreases. As a consequence, signals traveling through the upper atmosphere undergo severe non-linear distortion. This distortion takes two forms: waveform steepening and pulse stretching. These nonlinear effects are numerically investigated using non-linear ray theory. On one hand, waveform steepening is generally associated with stratospheric arrivals with sufficiently large amplitude for which the decreased density of the atmosphere causes shock fronts to form. On the other hand, pulse stretching is generally associated with thermospheric arrivals where severe attenuation prevents significant shock formation but severe non-linearity causes significant pulse stretching. Since non-linear effects increase with increasing signal amplitude, it is possible to use non-linear distortion to estimate signal source strength. Uncertainties in propagation path and in thermospheric attenuation will limit the accuracy of this approach. Using the non-linear propagation model, the fundamental limits of this approach are also investigated. Comparisons with available data will be made.

3:15

4pPA6. Pneumatic infrasound sources. Thomas Muir, Justin Gorhum, Charles Slack, Martin Barlett, and Timothy Hawkins (Applied Research Laboratories, University of Texas at Austin, P.O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu)

The generation of infrasound from the pulsation of compressed air is examined analytically and experimentally to explore the aerodynamic physics as well as engineering implementations. Several model experiments were developed and utilized to explore the problems associated with this approach. Applications to long range propagation in the atmosphere, including calibration and testing of infrasonic sensor systems are addressed. [Work supported by ARL:UT Austin.]

3:30

4pPA7. Application of a blind source separation algorithm for the detection and tracking of tornado-generated infrasound emissions during the severe weather outbreak of 27 April 2011. Hank S. Rinehart (Miltec Systems, Ducommun Miltec, 678 Discovery Dr NW, Huntsville, AL 35806, hrinehart@one.ducommun.com), Chris Clark, Matt Gray, and Kevin Dillion (Miltec Research and Technology, Ducommun Miltec, Oxford, MS)

April 25-28, 2011 has been identified by many as the most significant and severe single-system outbreak of tornadoes in recorded history. One day in particular, the 27th of April, has been classified by the National Oceanic and Atmospheric Administration (NOAA) as the fourth deadliest tornado outbreak in US history. Severe tornadic activity on this day levied catastrophic damage to life and property across areas of Mississippi, Alabama, Georgia and Tennessee. During this outbreak, multiple Ducommun Miltec-developed infrasound sensors collecting continuous, high resolution data were deployed in two-dimensional array configurations in Northern Alabama. Prior research on the collection and analysis of infrasonic emissions from severe weather phenomenon has provided much insight on the nature of tornado-generated infrasound. Our effort focuses on the application of novel bearing estimation algorithms using closely spaced (4-6 m) array elements. Direction of Arrival (DOA) estimates, derived from Blind Source Separation (BSS) techniques, will be presented for at least two significant tornadoes: the long-track EF5 that impacted Hackleburg and Phil Campbell, AL and the large multi-vortex EF4 that struck Cullman, AL. Correlation of infrasound detection and bearing estimate initiation and termination with NOAA Storm Prediction Center (SPC) Storm Reports will also be reviewed.

3:45

4pPA8. Infrasound as a remote sensing technique for the upper atmosphere. Jelle D. Assink, Roger Waxler, Joel Lonzaga, and Garth Frazier (NCPA/UM, 1 Coliseum Dr., (NCPA), University, MS 38677, jdassink@olemiss.edu)

Understanding and specification of the higher altitudes of the atmosphere with global overage over all local times is hampered by the challenges of obtaining direct measurements in the upper atmosphere. Methods to measure the properties of the atmosphere above the stratopause is an active area of scientific research. In this presentation, we revisit the use of infrasound as a passive remote sensing technique for the upper atmosphere. In the past, various studies focused on the sensitivity of infrasound to various upper atmospheric processes. It has been shown that the current state-of-the-art climatologies for the middle and upper atmosphere are not always in agreement with the acoustic data, suggesting a use of infrasound as a complementary remote sensing technique. Previously, we reported on the error in thermospheric celerities which was found to be in accord with the typical uncertainty in upper atmospheric winds and temperature. In this presentation, we report on the expected variation of the various infrasound observables from a forward modeling perspective. This information, in combination with the experimental measurement error provides constraints on the expected resolution from the inverse problem. With this information, we minimize misfits in travel time and source location using a Levenberg-Marquardt search algorithm in combination with ray theory.

Session 4pSA

Structural Acoustics and Vibration: Applications in Structural Acoustics and Vibration

Robert M. Koch, Cochair
Chief Technology Office, Naval Undersea Warfare Center, Newport, RI 02841-1708

R. Daniel Costley, Cochair
Geotechnical and Structures Lab., U.S. Army Engineer R&D Center, Vicksburg, MS 39180

Contributed Papers

3:30

4pSA1. Effects of acousto-optic diffraction in the acoustic frequency range on the laser Doppler vibrometry method in air. Hubert S. Hall, Joseph F. Vignola, John A. Judge, Aldo A. Glean, and Teresa J. Ryan (Mechanical Engineering, The Catholic University of America, 620 Michigan Ave NE, Washington, DC 20064, 61hall@cardinalmail.cua.edu)

The effect of acousto-optic diffraction on the transmitted laser signal light path in the laser Doppler vibrometry (LDV) method has been a known concern to measurement accuracy. To further understand and quantify the LDV accuracy implications, for the in-air case, an experimental study was performed in a high-intensity sound field within the acoustic frequency range (less than 20 kHz). Results of the study showed that acousto-optic diffraction has minimal impact on the accuracy of LDV measurements for the in-air case in sound-fields less than 20 kHz. Follow-on work investigating the effect of measurements in water is proposed. It is hypothesized that the higher refractive index of water will exacerbate the impact of the acousto-optic effect on the accuracy of LDV measurements. Previous work in the field in the megahertz frequency region has shown this. However, within the acoustic frequency range, the accuracy implications remain unknown.

3:45

4pSA2. References per coherence length: A figure of merit for multireference acoustical holography. Alan T. Wall, Michael D. Gardner, Kent L. Gee, and Tracianne B. Neilsen (Dept. of Physics and Astronomy, Brigham Young University, Provo, UT 84602, alantwall@gmail.com)

Multireference partial field decomposition (PFD) can be used to generate coherent holograms for scan-based near-field acoustical holography measurements. PFD is successful when the reference array completely senses all independent subsources, but meeting this requirement is not straightforward when the number of subsources and their locations are ambiguous (such as in aeroacoustic sources). A figure of merit based on spatial coherence lengths, called references per coherence length (RPLc), is a useful metric to guide inter-reference spacing in the array design so that the source is spanned. Coherence length is defined as the axial distance over which the ordinary coherence drops from unity to some desired value. Numerical experiments involving an extended, partially correlated source show that sufficiency of the reference array for different source conditions may be simply expressed in terms of RPLc. For sources of varying spatial coherence and over a large range of frequencies, one reference per coherence length is equivalent to sensing all independent subsources.

4:00

4pSA3. Physical quantities measurement by using the fiber optic sensor in the pendulum ball collision. Jongkil Lee (Mechanical Engineering Education, Andong National University, 388 Sonchun-dong, Andong, Kyungbuk 760-749, Republic of Korea, jlee@andong.ac.kr), Alex Vakakis (Mechanical Engineering, University of Illinois at Urbana-Champaign, Urbana, IL), and Larry Bergman (Aerospace Engineering, University of Illinois at Urbana-Champaign, Urbana, IL)

It is interesting to measure the impact force when the moving pendulum ball collides to the fixed body. In this paper physical quantities were measured using by fiber optic sensor when a pendulum ball collides to the fixed ball on the wall. Both steel ball dimensions are 1 inch in diameter. The fiber optic Sagnac interferometer is well established as a sensor for acoustic and vibration. It is made a 1mm penetrated hole and optical fiber in the Saganac loop passed through the hole. The ball was welded on the wall as a fixed ball. When the external force applied to the fixed ball the optical fiber in the Sagnac loop detects the impact force. The output signal is proportional to the output voltage in the oscilloscope. Based on the result suggested fiber optic sensor can measure the impact intensity and this technique can be expanded to the moving bodies.

4:15

4pSA4. Experimental investigation on reconstruction of sound field based on spherical near-field acoustic holography. Xinguo Qiu, Minzong Li, Huancai Lu, and Wei Jiang (Key Laboratory of E&M, Ministry of Education & Zhejiang Province, College of Mechanical Engineering, Zhejiang University of Technology, PO Box 21, 18 ChaoWang Road, Hangzhou, Zhejiang Province 310014, P.R.C., xinguochiu@126.com)

This paper presents the results of an experimental study on the methodology of spherical near-field acoustic holography (spherical NAH) to reconstruct interior sound field. The experiment was carried out in a full anechoic chamber, in which the sound filed was generated with different combination of speakers at different positions, a rigid spherical array was used to collect the field acoustic pressures as input to the reconstruction calculation. There are three cases which were investigated. Case 1, a source was set near to the microphone array. Case 2, two sources were eccentrically set opposite to each other around the microphone array. And Case 3, two sources were placed on one side of microphone array on the same orbit, while they were positioned apart at a small angle. The accuracy of the reconstruction of sound field was examined and analyzed compared to the benchmarks and the results of the numerical simulations. The reconstructed results show that the methodology of Spherical NAH is capable to locate sources and reconstruct sound field within certain accuracy.

Session 4pSC

Speech Communication: Speech Perception II: Intelligibility, Learning, and Audio-Visual Perception (Poster Session)

Michael S. Vitevitch, Chair
Psychology, University of Kansas, Lawrence, KS 66045

Contributed Papers

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

4pSC1. Confusability of Bengali consonants and its relation to phonological dissimilarity. Sameer ud Dowla Khan (Linguistics, Reed College, Portland, OR, sameeruddowlakhan@gmail.com)

Language-specific consonant similarity can be measured indirectly by looking at the phoneme inventory, the lexicon (e.g. cooccurrence restrictions), or the phonology (e.g. processes that take the notion of similarity of dissimilarity into account). A more direct approach involves the use of the confusion matrix. For Bengali, thus far, consonant similarity has only been measured indirectly, through the lexicon and phonology. Previous studies (Khan 2006, 2012) claim that Bengali speakers judge the similarity of consonants in echo reduplication (similar to English doctor-schmoctor), where the initial consonant of the base is systematically replaced with a phonologically dissimilar consonant in the reduplicant. This measurement of similarity assumes a set of features assigned language-specific weights; for example, [voice] is weighted more heavily that [spread glottis], to explain why speakers treat the pair [t, th] as more similar than the pair [t, d]. But does the measurement of similarity inherent in the echo reduplicative construction correspond to the relative perceptibility of different consonant contrasts? The current study compares the relative confusability of Bengali consonants produced in noise with the claims of phonological notions of similarity associated with echo reduplication.

4pSC2. Effects of age, hearing loss, and phonological neighborhood density on children's perceptual confusions. Mark Vandam (Boys Town National Research Hospital, 555 North 30th Street, Omaha, NE 68131, mrk.vandam@gmail.com), Noah H. Silbert (Center for Advanced Study of Language, University of Maryland, College Park, MD), and Mary Pat Moeller (Boys Town National Research Hospital, Omaha, NE)

Age, hearing loss, and phonological neighborhood density have been shown to substantially affect the accuracy of productions in a word imitation task [VanDam, et al., 161st ASA Meeting]. Older children (7 years of age) are more accurate than younger children (4 years of age), normal hearing children are more accurate than children with mild- to severe hearing loss, and words from sparse phonological neighborhoods are produced more accurately than are words from dense neighborhoods. In an ongoing series of analyses, we extend these findings by analyzing how patterns of perceptual confusion vary as a function of age, hearing status (normal hearing versus hearing loss), and phonological neighborhood structure. Multilevel cognitive models fit to confusion data provide detailed quantitative descriptions of perceptual space and response bias and enable analysis of between- and within-group variability. Results shed light on the organization of the lexicon in young children with both normal hearing and hearing loss, and add to our understanding of the relationship between speech production and speech perception in children.

4pSC3. Phonological neighborhood clustering coefficient influences word learning. Rutherford Goldstein and Michael S. Vitevitch (Psychology, University of Kansas, 1415 Jayhawk Blvd., Lawrence, KS 66045, mvitevit@ku.edu)

Network science is one approach used to analyze complex systems, and has been applied to a complex cognitive system, namely the phonological lexicon (Vitevitch, 2008). One of the measures provided by network science, termed the clustering coefficient or C, influences lexical processes such as speech production (Chan & Vitevitch, 2010) and speech perception (Chan & Vitevitch, 2009). The current study presents evidence of C influencing the process of learning new words. Participants were trained and tested on nonword-nonobject pairs over three lab sessions at one week intervals. Testing occurred immediately after training and after a one week interval. Participants were tested on a picture naming task, a two-alternative-forced-choice task, and a lexical decision task. Results show an advantage for learning new words with a high clustering coefficient. A spreading activation account is used to explain the findings.

4pSC4. Phonological neighborhood density and vowel production in children and adults. Benjamin Munson (Speech-Language-Hearing Sciences, University of Minnesota, 115 Shevlin Hall, 164 Pillsbury Drive SE, Minneapolis, MN 55455, munso005@umn.edu), Mary E. Beckman (Linguistics, Ohio State University, Columbus, Minnesota, MN), and Jan Edwards (Communicative Disorders, University of Wisconsin, Madison, WI)

Previous studies have shown that vowels in words with high phonological neighborhood densities (ND) are produced closer to the periphery of the vowel space than are vowels in low-ND words (Munson & Solomon, 2004; Scarborough, 2010; Wright, 2003). Different explanations for this phenomenon have been proposed. One hypothesis is that they reflect a speaker's attempt to maintain acoustic distinctiveness among similar-sounding words. If this were true, then we might expect that the effect of ND on vowel production would be smaller in children than in adults, given that children have overall smaller-sized lexicons than adults. To evaluate this, we examined the effect of ND on vowel production in children and adults. The productions were taken from the paidologos corpus (Edwards & Beckman, 2008). Preliminary analyses of the productions of 8 high-ND and 8 low-ND words by 10 2-year-olds, 10 5-year-olds and 20 adults have been completed. There are no effects of ND on vowel-space size or vowel duration for either group of children. There were strong, statistically significant effects found the group of adults. These were in the opposite than predicted direction: low-ND words were produced with more-expanded vowel spaces than high-ND words. Analysis of a larger group of participants is ongoing [support: NIDCD 02932.]

4pSC5. Optimized speech sound category training bootstraps foreign word learning. Han-Gyol Yi, Bharath Chandrasekaran (Communication Sciences and Disorders, The University of Texas at Austin, Austin, TX 78712, gyol@utexas.edu), and W. Todd Maddox (Psychology, The University of Texas at Austin, Austin, TX)

The COmpetition between Verbal and Implicit Systems (COVIS) model posits multiple cognitive learning systems that are functionally distinct and compete with each other throughout learning. COVIS indicates two cognitive systems: a hypothesis-testing system mediated predominantly by the frontal cortex and a procedural-based system mediated by the striatum. Initial learning is dominated by the hypothesis-testing system, but with increased practice, control is passed to the procedural system. Importantly, each learning system can be optimized differentially to maximize performance. In this study, the COVIS model was applied to optimize Mandarin tone learning by adult native English speakers. The optimized category training (OCT) was designed to boost the hypothesis-testing system initially, then the procedural-based system subsequently. We then examined the extent to which OCT enhanced performance in a word learning task which required the participants to use their attained categories for Mandarin tones to disambiguate words. OCT was found to significantly enhance word learning relative to a control condition in which the learning systems were not optimized. These results demonstrate that a multiple systems approach can be used to develop optimized training protocols to maximally boost category learning.

4pSC6. The young and the meaningless: Novel-word learning without meaning or sleep. Efthymia C. Kapnoula, Stephanie Packard, Keith S. Apfelbaum, Bob McMurray, and Prahlad Gupta (Psychology, The University of Iowa, Iowa City, IA 52242-1409, efthymiaevangelia-kapnoula@uiowa.edu)

Existing work suggests that sleep-based consolidation (Gaskell & Dumay, 2003) is required for newly learned words to interact with other words and phonology. Some studies report that meaning may also be needed (Leach and Samuel, 2007), making it unclear whether meaningful representations are required for such interactions. We addressed these issues by examining lexical competition between novel and known words during online word recognition. After a brief training on novel word-forms (without referents), we evaluated whether the newly learned items could compete with known words. During testing, participants heard word stimuli that were made by cross-splicing novel with known word-forms (NEP+NET=NEpT) and the activation of the target-word was quantified using the visual world paradigm. Results showed that the freshly learned word-forms engaged in competition with known words with only 15 minutes of training. These results are important for two reasons: First, lexical integration is initiated very early in learning and does not require associations with semantic representations or sleep-based consolidation. Second, given studies showing that lexical competition plays a critical role in resolving acoustic ambiguity (McMurray, Tanenhaus & Aslin, 2008; McMurray et al, 2009), our results imply that this competition does not have to be between semantically integrated lexical units.

4pSC7. Recognition memory in noise for speech of varying intelligibility. Rachael C. Gilbert (Linguistics, The University of Texas at Austin, Austin, TX 78751, rachaelgilbert@gmail.com), Bharath Chandrasekaran (Communication Sciences and Disorders, The University of Texas at Austin, Austin, TX), and Rajka Smiljanic (Linguistics, The University of Texas at Austin, Austin, TX)

Speech intelligibility is greatly affected by the presence of noise. However, there has been little research investigating the effects of noise on recognition memory. Per the effortfulness hypothesis (McCoy et al., 2005), we expect that processing speech in challenging listening environments requires additional processing resources that might otherwise be available for encoding speech in memory. This resource reallocation may be offset by speaker adaptations to the environment and to the listener. Here we compare recognition memory for conversational and clear speech sentences recorded in quiet (QS) and for sentences produced in response to the actual environment noise, i.e. noise adapted speech (NAS). Listeners heard 40 unique conversational and clear QS or NAS sentences mixed with 6-talker babble at SNRs of 0 or +3 dB. Following the exposure, listeners identified 80 sentences in quiet as old or new. Results showed that 1) increased intelligibility through conversational-to-clear speech modifications leads to improved recognition memory and 2) NAS presents a more naturalistic speech adaptation than QS, leading to better sentence recall for listeners. This experiment suggests that acoustic-phonetic modifications implemented in listener-oriented speech lead to improved speech recognition in challenging listening conditions and, crucially, to a substantial enhancement in recognition memory for sentences.

4pSC8. The intelligibility of clear and conversational allophones of coda consonants. Matthew J. Makashay, Nancy P. Solomon, and Van Summers (Audiology and Speech Center, Walter Reed National Military Medical Center, 8901 Wisconsin Ave., Bethesda, MD 20889-5600, matthew.j.makashay.civ@health.mil)

For many hearing-impaired (HI) listeners, hearing-aid amplification provides near-normal speech recognition in quiet. Nonetheless, many of these same listeners show large speech deficits, relative to normal-hearing (NH) listeners, that are not effectively addressed via amplification in noisy listening conditions. One compensating strategy HI listeners use is to ask talkers to speak clearly. However, as one of the features of clear speech is a shift to higher frequencies, HI listeners may not benefit as much as NH listeners if the new frequencies are outside their audible range. This study examined the intelligibility of conversationally- and clearly-spoken coda consonants in nonsense syllables. These free-variant allophones of 21 American English consonants were produced in three phonological environments: syllable (utterance) final; syllable final followed by schwa; and syllable final followed by palatal approximant and schwa. The stimuli were presented in broadband noise and in quiet to NH and HI listeners. Consonant confusions were investigated to determine whether NH and HI listeners receive similar clear-speech advantages. [The views expressed in this abstract are those of the authors and do not necessarily reflect the official policy or position of the Departments of the Navy, Army, or Air Force, the Department of Defense, or the US Government.]

4pSC9. Variability in speech understanding in noise by listeners with hearing loss. Peggy B. Nelson, Yingjiu Nie, Adam Svec, Tess Koerner, Bhagyashree Katare, and Melanie Gregan (University of Minnesota, 164 Pillsbury Dr Se, Minneapolis, MN 55455, peggynelson@umn.edu)

Listeners with sensorineural hearing loss (SNHL) report significant difficulties when listening to speech in the presence of background noise and are highly variable in their tolerance to such noise. In our studies of speech perception, audibility predicts understanding of speech in quiet for most young listeners with SNHL. In background noise, however, the speech recognition performance of some young listeners with SNHL deviates significantly from audibility predictions. We hypothesize that vulnerability to background noise may be related to listeners' broader auditory filters, to a loss of discrimination ability for rapid spectral changes, or to a disruption of the speech temporal envelopes by the addition of noise. Measures of spectral resolution, spectral change detection, and envelope confusion will be presented for listeners with SNHL. Relationships between those estimates and speech recognition in noise will be described. Results may suggest a range of custom strategies for improving tolerance for background noise. Work supported by NIDCD R018306 to the first author.

4pSC10. Relationships among word familiarity, volume unit level, root-mean-square power, and word difficulty. Edward L. Goshorn (Speech and Hearing Sciences, Psychoacoustics Research Laboratory, University of Southern Mississippi, 118 College Dr. #5092, Hattiesburg, MS 39401, edward.goshorn@usm.edu)

Due to time constraints, 10-25 item NU-6 word lists that are rank-ordered by word difficulty (W-D) are often used for audiological speech intelligibility testing rather than full 50-item lists (Hurley and Sell, 2003). The factors contributing to W-D are not well delineated. Although word familiarity is well established as an important contributor to W-D (Savin, 1963), the contributions of acoustical factors such as peak VU level (VU) and root-mean-square (RMS) power are less known. A better understanding of relationships among factors associated with W-D may prove useful in compiling word lists. This study investigated the relationships among word familiarity, VU, RMS power, and W-D for four 50-item NU-6 word lists. VU and RMS measures for each word were obtained with SoundForge. The standard frequency index (SFI) provided a measure of word familiarity (Carroll et al, 1971). An unexpected positive significant ($p<.05$) correlation was found between W-D and VU level for one list and weak positive correlations for three lists. Positive

correlations are in the opposite direction hypothesized for this variable. No significant correlations were found between W-D and RMS power. Significant negative correlations were found between W-D and SFI for three NU-6 lists. Expected and unexpected findings will be addressed.

4pSC11. Modeling talker intelligibility variation in a dialect-controlled corpus. Daniel McCloy, Richard Wright, and August McGrath (Linguistics, University of Washington, Box 354340, Seattle, WA 98115-4340, drmccloy@uw.edu)

In a newly created corpus of 3600 read sentences (20 talkers x 180 sentences), considerable variability in talker intelligibility has been found. This variability occurs despite rigorous attempts to ensure uniformity, including strict dialectal criteria in subject selection, speech style guidance with feedback during recording, and head-mounted microphones to ensure consistent signal-to-noise ratio. Nonetheless, we observe dramatic differences in talker intelligibility when the sentences are presented to dialect-matched listeners in noise. We fit a series of linear mixed-effects models using several acoustic characteristics as fixed-effect predictors, with random effects terms controlling for both talker & listener variability. Results indicate that between-talker variability is captured by speech rate, vowel space expansion, and phonemic crowding. These three dimensions account for virtually all of the talker-related variance, obviating the need for a random effect for talker in the model. Vowel space expansion is found to be best captured by polygonal area (contra Bradlow et al 1996), and phonemic overlap is best captured by repulsive force (cf. Liljencrants & Lindblom 1972, Wright 2004). Results are discussed in relation to prior studies of intelligibility.

4pSC12. The effects of prior access to talker information on vowel identification in single- and mixed-talker contexts. John R. Morton (Psychology, Washington University, Campus Box 1125, 1 Brookings Drive, Saint Louis, MO 63130, jrmhvc333@yahoo.com), Steven M. Lulich (Speech and Hearing Sciences, Indiana University, Saint Louis, MO), and Mitchell Sommers (Psychology, Washington University, Campus Box 1125, 1 Brookings Drive, Saint Louis, MO 63130)

Speech intelligibility is significantly impaired when words are spoken by multiple- compared with single talkers. In the present study, we examined whether providing listeners with information about the vocal tract characteristics of the upcoming speaker would reduce the difference between single- and mixed-talker conditions. All participants were initially trained to identify 6 talkers (3 male and 3 female) from isolated vowels. Participants then completed closed-set vowel identification tests, including a blocked and a mixed condition, and one of two mixed-talker precursor conditions. In one precursor condition, participants saw one of the six talker's name immediately prior to hearing the target vowel. In the other precursor condition, participants heard a non-target vowel (/i/) spoken immediately before the target stimulus. For both precursor conditions, the name (or vowel) precursor matched the target-vowel speaker on half of the trials. For the other half, the precursor was of the same gender as the target-vowel speaker. Only when a sample vowel precursor was spoken by the same talker as the subsequent target was there a significant improvement in scores relative to the mixed-talker condition. The results suggest that exposure to isolated vowels can provide enough information about that talker's vocal tract to improve perceptual normalization.

4pSC13. The relationship between first language and second language intelligibility in Mandarin-English bilinguals. Jenna S. Luque, Michael Blasingame, L. A. Burchfield, Julie Matsubara, and Ann R. Bradlow (Linguistics, Northwestern University, 2016 Sheridan Rd., Evanston, IL 60208, SLPJenna@gmail.com)

Previous research has shown that L1 speech intelligibility, as judged by native listeners, varies due to speaker-specific characteristics. Similarly, L2 speech intelligibility as judged by native listeners also varies across speakers. Given variability in L1 and L2 intelligibility, we hypothesize that, within bilinguals, some speaker-specific characteristics that contribute to variability in L1 intelligibility (e.g., long-term average spectrum, speech rate, and articulatory precision) are language-independent and therefore also contribute to variability in L2 intelligibility. This leads to the expectation that within a group of bilingual speakers, relative L1 intelligibility is a significant predictor of relative L2 intelligibility. In the current study, 14 Mandarin-English bilinguals produced 112 short meaningful sentences in their L1 (Mandarin) and L2 (English). Independent groups of Mandarin and English listeners then repeated back the sentences (native-accented Mandarin productions for Mandarin listeners at -4 dB SNR, Mandarin-accented English productionsfor English listeners at 0 dB SNR). Intelligibility was calculated as proportion of words correctly repeated. L1 intelligibility was a significant predictor of L2 intelligibility (β=.58, S.E.β=.21, $p<.05$) within these Mandarin-English bilinguals, supporting the hypothesis that language-independent speaker-specific characteristics contribute to both L1 and L2 intelligibility in bilingual speakers.

4pSC14. The role of first-language production accuracy and talker-listener alignment in second-language speech Intelligibility. Kyounghee Lee and Ann R. Bradlow (Department of Linguistics, Northwestern University, 2016 Sheridan Road, Evanston, IL 60208-4090, kyoungheelee2013@u.northwestern.edu)

This study investigated variability in L2 speech intelligibility as a function of L1 speech intelligibility and of talker-listener L1 match. Non-native Korean talkers varying in their L2 proficiency were recorded reading simple English (L2) and Korean (L1) sentences. The intelligibility of these sentences was then assessed by Korean listeners (both Korean and English productions) and English listeners (English productions only) in a sentence recognition task. The results revealed that for these Korean-English bilingual talkers, L1 intelligibility was significantly correlated with L2 intelligibility for both Korean and English listeners, suggesting that variability in L1 speech intelligibility can serve as a predictor of variability in L2 production accuracy. We also examined the interlanguage speech intelligibility benefit for non-native listeners (ISIB-L) (e.g., Bent & Bradlow, 2003; Hayes-Harb et al., 2008). Korean listeners performed better at identifying the sentences produced by Korean talkers with relatively low L2 intelligibility, implying that the benefit of a shared native language between talkers and listeners may be larger when non-native listeners process speech from a talker with low L2 intelligibility. Overall, these findings indicate that variability in L2 speech intelligibility is related to language-general talker characteristics as well as to the talker-listener language alignment.

4pSC15. The effect of age on phonetic imitation in children. Kuniko Nielsen (Linguistics, Oakland University, 320 O'Dowd Hall, Rochester, MI 48309-4401, nielsen@oakland.edu)

This study aims to examine the effect of age on phonetic imitation in children. Previous studies have shown that adult speakers implicitly imitate the phonetic properties of recently heard speech (e.g. Goldinger, 1998). Recently, Nielsen (2011) reported that third-graders show similar patterns of phonetic imitation, including word-specific patterns of imitation. The current study extends these findings and investigates the effect of age on phonetic imitation, by comparing the pattern of imitation between third-graders and preschoolers. According to Piaget (1962), development of imitation is a manifestation of the increasing distinctiveness between assimilation and accommodation in early childhood, predicting greater imitation for older children. At the same time, findings on motor imitation by newborns (e.g., Meltzoff & Moore, 1997) suggest that the intermodal mapping necessary for imitation is at least partly innate. The experiment employed a picture-naming task: participants' speech production was compared before and after they were exposed to model speech with extended VOT on the target phoneme /p/. A preliminary data analysis revealed a greater degree of imitation for older children, while both groups showed significant imitation. These findings are in agreement with the effects of age and developmental level on motor imitation observed in Fouts and Liikanen (1975).

4pSC16. Speech recognition informed by distinctive feature theory: The featurally underspecified Lexicon model and its implications. Philip J. Roberts (Faculty of Linguistics, University of Oxford, Clarendon Institute, Walton Street, Oxford OX1 2HG, United Kingdom, philip.roberts@ling-phil.ox.ac.uk) and Henning Reetz (Institut für Phonetik, Universität Frankfurt, Frankfurt, Hesse, Germany)

We present a speech recognition engine that implements the Featurally Underspecified Lexicon calculus (FUL). The FUL model defines an inventory of privative phonological features that is necessary and sufficient to describe contrasts between phonemes in any language in the world. The model also defines conditions for comparing feature bundles recovered from the signal with segments defined in the lexicon: a feature may MATCH (the feature is present in both the signal and the lexicon), it may MISMATCH (the feature in the signal is impossible in tokens of the segment in the lexicon, e.g. when a stop-burst, which indicates the [PLOSIVE] feature, is compared with a segment carrying the [CONTINUANT] feature in the lexicon), or it may provoke a NOMISMATCH response when the feature in the signal is not part of the segment in the lexicon. This matching mechanism also accounts for asymmetries as they are observed in natural speech, as [CORONAL] assimilates to [LABIAL] but not the other way around. The engine computes distances to neighboring words according to a coherence measure to simulate co-activation in the lexicon. We will demonstrate online the operation of this engine in English and German.

4pSC17. Evaluating automatic speech-to-speech interpreting. Jared Bernstein (Linguistics, Stanford University, Palo Alto, CA 94301, jared413@stanford.edu) and Elizabeth Rosenfeld (Tasso Partners, Palo Alto, CA)

A naïve speech-to-speech interpreter can be implemented as three component processes in series: speech recognition, machine translation, and speech synthesis. However, evaluating the performance of a speech-to-speech interpreting system may be either simpler or more complicated than merely calculating the product of the accuracies of those three component processes. This is because human users are sensitive to the rate at which an interpreter operates and because a system's communication success rate is properly measured by a listener's correct understanding of the speaker's intention in a particular context rather than by the system's word-for-word accuracy and intelligibility. We are evaluating two currently available systems, each operating in both directions for two language pairs: English/Spanish and English/Chinese. For each system and each language pair and each direction, we compare word-for-word spoken interpretation accuracy with the successful communication of speaker intent-in-context. Results for the Spanish/English pair suggest that word-for-word accuracy is high (about 75-90% correct) for both systems, and that taking a lower information rate measure like communication of intent in context reduces the error rate substantially. Finally, suggestions for improved system design are presented.

4pSC18. Parametric forms of spectral amplitude nonlinearities for use in automatic speech recognition. Stephen Zahorian (Electrical and Computer Engineering, State University of New York at Binghamton, PO Box 6000, Binghamton, NY 13902, zahorian@binghamton.edu)

This work is a continuation and extension of work presented at the fall 2011 meeting of the Acoustical Society of America (Wong and Zahorian). In that work, and also at work done at Carnegie Mellon University, auditory model derived spectral amplitude nonlinearities, with symmetric additional compression (after log amplitude scaling) were found to improve automatic speech recognition performance when training and test data are mismatched with respect to noise. In this new work, a parametric nonlinearity, controlled by three parameters, was formed to allow non-symmetric compression with respect to high and low amplitudes. Several variations of this basic nonlinearity were evaluated for both matched and mismatched conditions for training and test data with respect to noise. For the mismatched cases, the most effective nonlinearity was found to be compressive for low amplitudes but expansive at high amplitudes (thus emphasizing spectral peaks). However, for matched conditions, none of the spectral amplitude nonlinearities improve automatic recognition accuracy. The additional nonlinearity can be combined with log compression to create a single unified amplitude nonlinearity for speech processing.

4pSC19. Acoustical features in Mandarin emotional speech by native speakers of English. Hua-Li Jian (Fac. Tech. Des. and Art, Oslo and Akershus University College of Applied Sciences, Postbox 4, St. Olavs Plass, Oslo NO-0130, Norway, Hua-Li.Jian@hioa.no) and Jessica Hung (Foreign Languages and Literature, National Cheng Kung University, Tainan, Tainan City, Taiwan)

This study examines (1) whether native speakers of English (NS-E) can express emotions successfully in Mandarin speech, and (2) how their emotional expressions differ from native speakers of Mandarin (NS-C) when the emotional portrayals are recognizable. The acoustic features analyzed included F0, duration, and intensity. The scenario approach was adopted to elicit emotions joy, anger, sadness and fear, with neutral as a control. The data gathered (Sixteen NS-E and NS-C) were rated. F0 range at sentential level, mean F0 of each syllable, sentential and syllabic duration, and intensity signal of each segment were contrasted across groups within each emotional expression. The findings indicated that emotions by NS-C were recognized well, but joy, anger and fear by NS-E had low recognition rates because of accents, vocal cues and culture-specific components. Both groups adopted similar F0 range at the sentence level but joy in both and fear by NS-C showed small range. Only NS-C showed fast speech rate in anger. Emotions with high activations by NS-E were shorter. Anger and joy showed high intensity, while sadness and fear low intensity in both groups. NS-C tended to use different intensity range to indicate different emotions, while NS-E used similar range for all emotions.

4pSC20. Cross-linguistic emotion recognition: Dutch, Korean, and American English. Jiyoun Choi, Mirjam Broersma (Max Planck Institute for Psycholinguistics, PO Box 310, Nijmegen 6500 AH, Netherlands, mirjam.broersma@mpi.nl), and Martijn Goudbeek (Tilburg University, Tilburg, n/a, Netherlands)

This study investigates the occurrence of asymmetries in cross-linguistic recognition of emotion in speech. Theories on emotion recognition do not address asymmetries in the cross-linguistic recognition of emotion. To study perceptual asymmetries, a fully crossed design was used, with speakers and listeners from two typologically unrelated languages, Dutch and Korean. Additionally, listeners of American English, typologically close to Dutch but not Korean, were tested. Eight emotions, balanced in valence (positive-negative), arousal (active-passive), and basic vs. non-basic emotions -properties that are known to affect emotion recognition- were recorded by eight Dutch and eight Korean professional actors, in a nonsense phrase that was phonologically legal in both languages (and English). Stimuli were selected on the basis of prior validation studies with Dutch and Korean listeners. 28 Dutch, 24 Korean, and 26 American participants were presented with all 256 Dutch and Korean stimuli, blocked by language. Participants indicated for each stimulus which emotion it expressed by clicking on one of the eight emotions or "neutral". Results showed strong asymmetries across languages and listener groups that cannot be explained along previously described dimensions (valence, arousal, basic-non-basic). The present results call for the extension of theories of cross-linguistic emotion recognition to incorporate asymmetrical perception patterns.

4pSC21. Cues for the perception of expressive speech. Daniel J. Hubbard and Peter F. Assmann (School of Behavioral and Brain Sciences, GR4.1, University of Texas at Dallas, P.O. Box 830688, Richardson, TX 75083, dhubbard@utdallas.edu)

In a previous study the contribution of fundamental frequency (F0) to the perception of expressive speech was examined using a selective adaptation technique. Listeners heard either F0-present or F0-removed adaptors (vocoder-processed VCV syllables) from one of two expressive categories: angry or happy. In a two-alternative listening task, contrastive aftereffects (characterized by a tendency to label test stimuli as originating from the non-adapted category) were documented only in the F0-present condition. This suggests that F0 is important for the perception of emotional expressions. However, listeners were still able to identify F0-removed syllables as angry or happy at a rate significantly better than chance (58%). Subsequent analyses revealed systematic differences in formant frequencies as potential cues for categorization (higher F1 and F2 frequencies for happy compared to angry tokens). A discriminant analysis was performed using formant measurements (F1-F3) and measures related to the voicing source. Removing voice source measures (including mean F0) produced a decrease in classification accuracy that closely matched listener response patterns for the F0-removed stimuli. The results suggest that in the absence of F0, formant frequencies may be used for perception of angry and happy speech.

4pSC22. Perception of speaker sex and vowel recognition at very short durations. David R. Smith (Psychology, University of Hull, Cottingham Road, Hull HU6 7RX, United Kingdom, d.r.smith@hull.ac.uk)

A man or woman saying the same vowel do so with very different voices. The auditory system solves the problem of extracting what the man or woman said despite substantial differences in the acoustic properties of the carrier voice. Much of this acoustic variation is due to differences in the underlying anatomical mechanisms for producing speech. If the auditory system knew the sex of the speaker then it could correct for speaker-sex related acoustic differences thus facilitating vowel recognition. We measured the minimal stimulus duration necessary to accurately discriminate whether a brief vowel segment was spoken by a man or woman, and to accurately recognize what vowel was spoken. Results show that reliable vowel recognition precedes reliable speaker sex discrimination. Furthermore, the pattern of performance across experiments where voice pitch and resonance information were systematically varied, is markedly different depending on whether the task is speaker-sex discrimination or vowel recognition. These findings suggest that knowledge of speaker sex has little impact upon vowel recognition at very short stimulus durations.

4pSC23. Partial effects of perceptual compensation need not be auditorily driven. Gregory Finley (Linguistics, University of California, Berkeley, CA 94720, finley@berkeley.edu)

An experiment was devised to test whether compensation for coarticulation could be motivated by nonspeech for which gestural recovery is impossible. Subjects were presented with CV stimuli formed by concatenating an /s ~ ʃ/ continuum fricative with an /i/ or /o/ from three types of synthesized vocalic nuclei: full spectral vowels (Set A), vowels with F2 but no other formants (B), and pure sine tones at F2 (C). F2 was chosen as the common parameter between sets because extremely high or low F2 is enough information for English speakers to judge vowel roundedness (and rounding lowers centroid frequency, a key difference between /s/ and /ʃ/). Comparing fricative identification boundaries between the vowels within each set, compensation occurred reliably in Sets A ($t = 4.6$, $p < 0.01$) and B ($t = 3.7$, $p < 0.01$) but not C. Comparing conditions against each other, Set A showed significantly higher vowel-triggered boundary shift than B ($F = 8.18$, $p < 0.05$). Results support the conclusion that relevant acoustic cues might not trigger compensation if they cannot be associated with speech; additionally, sounds reminiscent of speech do not generate as strong an effect as sounds firmly identifiable as speech.

4pSC24. Language specific compensation for coarticulation. Keith Johnson, Shinae Kang, Greg Finley, Carson Miller Rigoli (Linguistics, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, keithjohnson@berkeley.edu), and Elsa Spinelli (Psychology and Neurocognition, Université Pierre Mendès-France, Grenoble, Cedex 9, France)

This paper reports an experiment testing whether compensation for coarticulation in speech perception is mediated by linguistic experience. The stimuli are a set of fricative-vowel syllables on continua from [s] to [ʃ] with the vowels [a], [u], and [y]. Responses from native speakers of English and French (20 in each group) were compared. Native speakers of French are familiar with the production of the rounded vowel [y] while this vowel was unfamiliar to the native English speakers. Both groups showed compensation for coarticulation (both $t > 5$, $p < 0.01$) with the vowel [u] (more "s" responses indicating that in the context of a round vowel, fricatives with a lower spectral center of gravity were labeled "s"). The French group also showed a compensation effect in the [y] environment ($t[20] = 3.48$, $p<0.01$). English listeners also showed a tendency for more subject-to-subject variation on the [y] boundary locations than did the French listeners (Levene's test of equality of variance, $p < 0.1$). The results thus indicate that compensation for coarticulation is a language specific effect, tied to the listener's experience with the conditioning phonetic environment.

4pSC25. Audio/visual compensation for coarticulation. Shinae Kang, Greg Finley, Keith Johnson, and Carson Miller Rigoli (Linguistics, UC Berkeley, UC Berkeley, Berkeley, CA 94720-2650, sakang2@berkeley.edu)

This study investigates how visual phonetic information affects compensation for coarticulation in speech perception. A series of CV syllables with fricative continuum from [s] to [sh] before [a],[u] and [y] was overlaid with a video of a face saying [s]V, [ʃ]V, or a visual blend of the two fricatives. We made separate movies for each vowel environment. We collected [s]/[ʃ] boundary locations from 24 native English speakers. In a test of audio-visual integration, [ʃ] videos showed significantly lower boundary locations (more [sh] responses) than [s] videos ($t[23]=2.9$, $p<0.01$) in the [a] vowel environment. Regardless of visual fricative condition, the participants showed a compensation effect with [u] ($t[23] > 3$, $p<0.01$), but not with the unfamiliar vowel [y]. This pattern of results was similar to our findings from an audio-only version of the experiment, implying that the compensation effect was not strengthened by seeing the lip rounding of [y].

4pSC26. The influence of visual information on the perception of Japanese-accented speech. Saya Kawase, Beverly Hannah, and Yue Wang (Department of Linguistics, Simon Fraser University, Burnaby, BC V5A 1S5, Canada, skawase@sfu.ca)

This study examines how visual information in nonnative speech affects native listener judgments of second language (L2) speech production. Native Canadian English listeners perceived three English phonemic contrasts (/b-v, θ-s, l-ɹ/) produced by native Japanese speakers as well as native Canadian English speakers as controls. Among the stimuli, /v, θ, l, ɹ/ are not existent in the Japanese consonant inventory. These stimuli were presented under audio-visual (AV), audio-only (AO), and visual-only (VO) conditions. The results showed that while overall perceptual judgments of the nonnative phonemes (/v, θ, l, ɹ/) were significantly less intelligible than the native phonemes (/b,s/), the English listeners perceived the Japanese productions of the phonemes /v, θ, b,s/ as significantly more intelligible when presented in the AV condition compared to the AO condition. However, the Japanese production of /ɹ/ was perceived as less intelligible in the AV compared to the AO condition. Further analysis revealed that a significant number of Japanese productions of /ɹ/ lacked lip-rounding, indicating that nonnative speakers' incorrect articulatory configurations may decrease intelligibility. These results suggest that visual cues in L2 speech productions may be either facilitative or inhibitory in native perception of L2 accented-speech. [Research supported by SFU and SSHRC.]

4pSC27. Effects of visual cue enhancement on speech intelligibility for clear and conversational speech in noise. Jasmine Beitz, Kristin Van Engen (Communication Sciences and Disorders, University of Texas at Austin, 1 University Station, Austin, TX 78712, jas.speechpath@gmail.com), Rajka Smiljanic (Linguistics, University of Texas at Austin, Austin, TX), and Bharath Chandrasekaran (Communication Sciences and Disorders, University of Texas at Austin, 1 University Station A1100, College of Communication, Austin, TX 78712)

Visual presentation of a speaker enhances the auditory perception of speech information in noisy conditions (e.g. Helfer and Freyman, 2005; Helfer, 1997). Intelligibility is also improved when a speaker adopts a clear speaking style (Smiljanic and Bradlow, 2009). The present study investigates the contributions of these two intelligibility-enhancing factors in the presence of several types of noise, which vary with respect to the degree of informational and energetic masking they impose on target speech. Specifically, it measures sentence intelligibility in the presence of 1 competing talker, 2-talker babble, 4-talker babble, and speech-shaped noise. Meaningful sentences (in clear and conversational styles) were presented to participants in each modality (audio-only; audio-visual) and in all noise conditions. Participants reported all intelligible words. Our data shows overall better intelligibility for clear speech and for AV speech relative to conversational speech and audio-only condition. However, the visual benefit associated with conversational speech is significantly greater than the visual benefit associated with clear speaking style. The relative contribution of visual influences and speech clarity to intelligibility enhancements will be discussed.

4pSC28. Sumby and Pollack revisted: The influence of live presentation on audiovisual speech perception.. Justin M. Deonarine, Emily J. Dawber, and Kevin G. Munhall (Psychology, Queen's University, Humphrey Hall, Queen's University, Kingston, ON K7L 3N6, Canada, justin.deonarine@yahoo.com)

In their classic paper, Sumby and Pollack (1954) demonstrated that the sight of a talker's face enhanced the identification of auditory speech in noise. Recently, there has been interest in the influence of some of their methodologies on audiovisual speech perception. Here, we examine the effects of presenting the audiovisual stimuli live, like Sumby and Pollack. Live presentation yields 3D visual stimuli, higher resolution images, and social conditions not present in modern replications with recorded stimuli and display monitors. Subjects were tested in pairs and alternated in the same session between a live talker (Live Condition) and a live feed of the talker to a television screen (Screen Condition). Order of presentation mode and word lists (monosyllabic English words) were counterbalanced across subjects. Subjects wore sound isolating headphones and signal intensity was controlled across conditions. Word lists were counterbalanced for spoken word frequency and initial consonant structure. Stimuli were presented in 7 signal-to-noise ratios (pink noise). Accuracy of identification was higher in the Live Condition than the Screen Condition. Possible causes of this effect are explored through manipulations of monocular and binocular depth cues and through testing with modern 3D display technology.

THURSDAY AFTERNOON, 25 OCTOBER 2012 MARY LOU WILLIAMS A/B, 2:00 P.M. TO 5:00 P.M.

Session 4pUW

Underwater Acoustics and Signal Processing in Acoustics: Array Signal Processing and Source Localizations

Jeffrey A. Ballard, Chair
Applied Research Laboratories, The University of Texas, Austin, TX 78713-8029

Contributed Papers

2:00

4pUW1. Performance of a large-aperture array in a complex littoral environment. Steven L. Means, Stephen C. Wales, and Jeffrey S. Rogers (Naval Research Laboratory, 4555 Overlook Ave SW, Washington, DC 20375, steven.means@nrl.navy.mil)

Over 850 hours, from an ~2 month time period, of ambient acoustic measurements were taken on a long (917m), 500 phone linear array 12 km off the coast of Fort Lauderdale, Florida, capturing both day and night, commercial and recreational shipping generated noise. Marine-band radar and AIS data were collected concurrently so that ship locations could be tracked in a large area surrounding the array. Array performance is investigated by beamforming in a number of frequency bands and apertures to determine median beam noise, noise gain reduction, and noise window statistics as a function of bearing. Additionally, the mean-square coherence is computed as a function of normalized distance (range/wavelength). The results are compared for a variety of time periods and environmental conditions. Comparisons of measurements with a computational noise model using known ship locations will be made. (Work supported by ONR base funding at NRL.)

2:15

4pUW2. Depth-based suppression of moving interference with vertical line arrays in the deep ocean. Reid K. McCargar and Lisa M. Zurk (Electrical and Computer Engineering, Northwest Electromagnetics and Acoustics Research Lab, Portland State University, PO Box 751, Attn NEAR Lab, Portland, OR 97207, rmccar@pdx.edu)

Vertical line arrays (VLAs) deployed below the critical depth in deep ocean environments can exploit the reliable acoustic path (RAP), which exhibits low transmission loss (TL) at moderate ranges and increased TL for distant interference. However, nearby surface ship interference presents a major challenge for an array lacking horizontal aperture that doesn't provide bearing discrimination. The motion of the interference degrades covariance estimation and limits observation intervals, thus limiting adaptive rejection capabilities. An image-method interpretation of the propagation physics reveals a depth-dependent modulation feature which enables separation of signals originating from near-surface and those from sub-surface passive acoustic sources. This discrimination can be achieved in the data through an integral transform. The feature is robust to environmental variability and allows for rejection of near-surface interference and depth classification. The transform-based filter is derived in closed form, and demonstrated with simulation results for a deep ocean environment with multiple moving surface interferers.

2:30

4pUW3. Imaging objects with variable offset in an evanescent wave field using circular synthetic aperture sonar and spectroscopy. Daniel Plotnick and Philip L. Marston (Dept. of Physics and Astronomy, Washington State University, Pullman, WA 99163, dsplotnick@gmail.com)

Imaging properties of objects suspended in an acoustic evanescent wave field are examined. Evanescent waves are generated using a tank containing immiscible liquids and an appropriately directed acoustic beam [C. F. Osterhoudt et al., IEEE J. Oceanic Eng. 33, 397-404 (2008)]. The source and receiver transducers are in the liquid having the higher sound velocity. Object(s) are spun about a vertical axis while scattering is measured. The object(s) offset into the wave field is then varied and the experiment repeated. In this work small spheres and other objects are used to gain insight into imaging properties as a function of the object or object(s) position in the evanescent field. Data is examined using circular synthetic aperture techniques. Additionally, a comparison is made between spectral data and a heuristic model in the case of two spheres. The spectral evolution in that case is affected by the interference from the two scatterers. Cases where the source and receiver are collocated (monostatic) and those where the source and receiver are separated (bistatic) are compared. [Work supported by ONR.]

2:45

4pUW4. Comparison of compressive sampling beamforming to adaptive and conventional beamforming. Jeffrey A. Ballard (Applied Research Laboratories, The University of Texas, P.O. Box 8029, Austin, TX 78713-8029, ballard@arlut.utexas.edu)

Interest in the application of compressive sampling to beamforming has increased in recent years [G.F. Edelmann and C.F. Gaumond, J. Acoust. Soc. Am. 130(4) EL232-EL237 (2011)]. This work compares the performance of a compressive sampling beamformer to conventional beamforming and to an adaptive beamforming algorithm which utilizes dominant mode rejection. The algorithms are applied to measured data collected on a horizontal line array off the southeast coast of Florida in 2008 [K. D. Heaney and J. J. Murray, J. Acoust. Soc. Am. 125(3), 1394-1402 (2009)]. Several factors affecting performance are considered: high and low SNR signals, amount of clutter (number of other targets), and varying degrees of array health (force failing 50%, 75%, & 90% of the array). To assess the ability of each beamformer to differentiate signals from noise, known signals are injected into the data at decreasing SNR. The analysis compares the frequency-azimuth data and bearing-time-records that each beamformer outputs. [Work supported by ARL:UT IRD.]

3:00

4pUW5. Frequency-difference beamforming with sparse arrays. Shima H. Abadi (Mechanical Engineering, University of Michigan, 2010 W.E.Lay Automotive Laboratory 1231 Beal Ave., Ann Arbor, MI 48109, shimah@umich.edu), Heechun Song (Marine Physical Laboratory, Scripps Institution of Oceanography, University of California at San Diego, La Jolla, CA), and David R. Dowling (Mechanical Engineering, University of Michigan, Ann Arbor, MI)

When an acoustic signal is transmitted to a remote receiving array with sufficient aperture and transducer density, the arrival direction(s) of the ray paths linking the source and the array may be determined by beamforming the transducer recordings. However, when the receiving array is sparse, i.e. there are many signal wavelengths between transducers, the utility of conventional beamforming is degraded because of spatial aliasing. Yet, when the signal has sufficient bandwidth, such aliasing may be mitigated or eliminated through use of an unconventional nonlinear beamforming technique that manufactures a desired frequency difference from the recorded signals. When averaged through the signal's frequency band, the output of frequency-difference beamforming is similar to that of conventional beamforming evaluated at the desired difference frequency. Results and comparisons from simple propagation simulations and FAF06 experimental measurements are shown for broadband signal pulses (11-19 kHz) that propagate 2.2 km underwater to a vertical 16-element receiving array having a 3.75-m-spacing between elements (almost 40 signal-center-frequency wavelengths). Here, conventional delay-and-sum beamforming results in the signal's frequency band are featureless, but received ray-path directions are successfully determined using frequency differences that are well below the broadcast signal's frequency band. [Sponsored by ONR.]

3:15–3:30 Break

3:30

4pUW6. Blind source localization and separation in three-dimensional space. Na Zhu (Department of Engineering Technology, Austin Peay State University, P.O. Box 4455, Clarksville, TN 47044, zhun@apsu.edu) and Sean Wu (Department of Mechanical Engineering, Wayne State University, Detroit, MI)

A new methodology called blind source localization and separation (BSLS) is developed to locate sound sources in three-dimensional space and extract their corresponding sound signals from the directly measured data. This technology includes two steps: Firstly, locate the sound sources by applying signal pre-processing and triangulation algorithms to the signals measured at microphones. Secondly, taking the sound source location results from the first step and measured signals at microphone as the input data, use the point source separation method to extract the sources and reconstruct the sound sources at their locations. The impact of various factors, such as the types and characteristics of sources, microphone configurations, signal to noise ratios, number of microphones, and errors in source localizations on the quality of source separation will be examined and compared to those obtained by the conventional blind source separation method.

3:45

4pUW7. Broadband sparse-array blind deconvolution using unconventional beamforming. Shima H. Abadi (Mechanical Engineering, University of Michigan, 2010 W.E.Lay Automotive Laboratory 1231 Beal Ave., Ann Arbor, MI 48109, shimah@umich.edu), Heechun Song (Marine Physical Laboratory, Scripps Institution of Oceanography, University of California at San Diego, La Jolla, CA), and David R. Dowling (Mechanical Engineering, University of Michigan, Ann Arbor, MI)

Sound from a remote underwater source is commonly distorted by multipath propagation. Blind deconvolution is the task of estimating the unknown waveforms for the original source signal, and the source-to-receiver impulse response(s), from signal(s) recorded in an unknown acoustic environment. Synthetic time reversal (STR) is a technique for blind deconvolution of underwater receiving-array recordings that relies on generic features of the propagating modes or ray paths that lead to multipath sound propagation. In prior studies the pivotal ingredient for STR, an estimate of the source-signal's phase (as a function of frequency), was generated from conventional beamforming of the recorded signals. However, through the use of unconventional nonlinear frequency-difference beamforming, STR can be extended to sparse array recordings where the receiving-array elements are many wavelengths apart and conventional beamforming is inadequate. This extension of STR was tested with simple propagation simulations and FAF06 experimental measurements involving broadband signal pulses (11-19 kHz) that propagate 2.2 km in the shallow ocean to a vertical 16-element receiving array having a 3.75-m-spacing between elements (almost 40 signal-center-frequency wavelengths). The cross-correlation coefficient between the source-broadcast and STR-reconstructed-signal waveforms for the simulations and experiments are 98% and 91-92%, respectively. [Sponsored by ONR.]

4:00

4pUW8. Beam steering response matrix inversion method: Accurate localization of simultaneous high and low frequency sources using a small line array. Jon W. Mooney (Acoustics & Vibration, KJWW Engineering Consultants, 623 26th Avenue, Rock Island, IL 61201, mooneyjw@kjww.com)

The Beam Steering Response (BSR) Matrix Inversion Method is a technique using a small line array that, when mathematically steered to populate a BSR matrix, can simultaneously resolve the locations of multiple sources. Using this technique, high directivity over a wide frequency range may be achieved with a small array. Simultaneous tracking of multiple low and high frequency sources is demonstrated using the BSR Matrix Inversion Method with an acoustic video camera having a 49 inch, 8 element line array. This paper addresses 1) the derivation of the Beam Steering Response matrix, 2) line array design considerations, 3) processing limitations, and 4) a practical demonstration of the BSR Matrix Inversion Method.

4:15

4pUW9. An experimental study of drive-by single channel time reversal using scaled targets and clutter. Ahmad T. Abawi (HLS Research, 3366 North Torrey Pines Court, Suite 310, La Jolla, CA 92037, abawi@hlsresearch.com), Ivars Kirsteins (NUWC, Newport, RI), and Daniel Plotnick (Physics, Washington State University, Pullman, WA)

Previous results suggest that iterative single channel time reversal (TR) is a promising, simple and inexpensive technique to enhance the backscatter signature of elastic objects and to simultaneously focus on its dominant resonance frequency in the presence of noise and clutter (Pautet et al. 2005), (Waters et al. 2009). However, to the best of our knowledge, these and other studies have only considered the case when the sonar system, target, and clutter are fixed or stationary, i.e., not moving. This is an unrealistic assumption since the sonar platform is typically moving. In fact, motion and environmentally-induced clutter scintillations may actually be beneficial for TR when the dominant target resonance varies slowly with aspect in clutter-limited noise environments. Theoretical arguments in (Kirsteins et al. 2009) suggest TR provides additional signal-to-noise ratio (SNR) enhancements against clutter under these circumstances. To confirm this hypothesis, a drive-by TR experiment was conducted in a tank using scaled elastic targets/clutter sources with the platform motion simulated by rotating the target/clutter during each TR iteration. Quantitative analysis in regards to detecting the dominant resonance clearly shows that TR in drive-by mode provides SNR enhancements against clutter.

4:30

4pUW10. Estimation of motion parameters and trajectory of an low flying aircraft using acoustic sensors. A. Saravanakumar (MIT Campus, Annauniversity, Chrompet, Chennai, Tamilnadu 600044, India, saravanakumar_a@yahoo.com)

An aircraft generates an acoustic impulse that propagates outwards from the source. The position of the source and hence the trajectory can be estimated by measuring the relative time of arrival of the impulse at a number of spatially distributed sensors. The time difference for the acoustic wave front to arrive at two spatially separated sensors is estimated by cross correlating the digitized outputs of the sensors. The time delay estimate is used to calculate the source bearing and the position of the source is found using triangulation technique using the bearings from two widely separated receiving nodes. The flight parameter of the aircraft is obtained by autocorrelation method using acoustic Multipath delays. The signal emitted by an UAV arrives at a stationary sensor located above a flat ground via a direct path and a ground-reflected path. The difference in the times of arrival of the

direct path and ground-reflected path signal components is known as the Multipath delay. A model is developed to predict the variation with time and autocorrelation method is formulated to estimate the motion parameters like speed and altitude of the aircraft.

4:45

4pUW11. Accuracy of passive source localization approaches with a single towed horizontal line-array in an ocean waveguide. Zheng Gong (Mechanical Engineering, Massachusetts Institute of Technology, 5-435, 77 Massachusetts Ave, Cambridge, MA 02139, zgong@mit.edu) and Purnima Ratilal (Electrical and Computer Engineering, Northeastern University, Boston, MA)

Instantaneous passive source localization applying the (1) synthetic aperture tracking, (2) array invariant, (3) bearings-only target motion analysis in modified polar coordinates via the extended Kalman filter, and (4) bearings-migration minimum mean-square error methods to measurements made on a single towed horizontal receiver array in a random range-dependent ocean waveguide are examined. These methods are employed to localize and track a vertical source array deployed in the far-field of a towed horizontal receiver array during the Gulf of Maine 2006 Experiment. The source transmitted intermittent broadband pulses in the 300-1200 Hz frequency range. All four methods are found to be comparable with averaged error of between 9% to 13% in estimating the mean source position in a wide variety of source-receiver geometries. In the case of a relatively stationary source, the synthetic aperture tracking outperforms the other three methods by a factor of two with only 4% error. For a moving source, the Kalman filter method yields the best performance with 8% error. The array invariant is the best approach for localizing sources within the endfire beam of the receiver array with less than 10% error.

THURSDAY EVENING, 25 OCTOBER 2012 — 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees on the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings 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 Thursday are as follows:

Animal Bioacoustics	Julia Lee A/B
Noise	Trianon C/D
Speech Communication	Mary Lou Williams
Underwater Acoustics	Bennie Moten

Session 5aAA

Architectural Acoustics and Noise: Potpourri

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

Contributed Papers

8:00

5aAA1. Defusing the controversy of scattering versus diffusion coefficients. Hendrik D. Gideonse (Sound Recording Technology, U. Mass Lowell, 294A Boston Avenue, Medford, MA 02155, hendrikxix@xix-acoustics.com)

Several methodologies for the testing and analysis of diffusers have been developed including the ISO Scattering Coefficient and the AES Diffusion Coefficient. These coefficients are the source of some controversy today and this paper makes the attempt to investigate the benefits and weaknesses of these tools by using them to research and test a new diffuser shape. Several issues are exposed in using the coefficients as both qualitative and quantitative metrics. The most important of these being problems with the validity of the comparison of the diffuser's behavior to that of a like-sized flat panel. Additionally there appears to be an intuitive disconnect between the perceived diffusive merits shown by polar plots and the numerical value of coefficients derived from the plots.

8:15

5aAA2. Noise mitigation at the Combat Arms Training Facility, Wright Patterson Air Force Base, Dayton, OH. William J. Murphy, Edward L. Zechmann, Chucri A. Kardous (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 4676 Columbia Parkway, Cincinnati, OH 45226-1998, wjm4@cdc.gov), and Ning Xiang (School of Architecture, Rensselaer Polytechnic Institute, Troy, NY)

The Combat Arms Training Facility (CATF) at Wright Patterson Air Force Base in Dayton Ohio was evaluated for the effect of noise treatment to the interior of the firing range. Measurements were conducted in 2009 and 2010 before and after noise treatment. Reverberant energy in the range was reduced through the installation of cementatious shredded fiber board and basalt rock wool to the walls and ceiling of the range. No modifications were made to the windows and doors connecting the range interior to adjacent rooms. Prior to the application of noise treatment, the reverberation times (RT60) ranged from about 3.5 seconds at 100 Hz to about 1.3 seconds at 10 kHz. Following application of the noise treatments, the RT60 was reduced to about 1.6 seconds at 100 Hz to 0.5 seconds at 10 kHz. The critical distance for speech intelligibility increased from about 12 feet to about 22 feet in the speech frequencies 800 to 4000 Hz. The Articulation loss of consonants was improved from 22.5 for the untreated range to 7.2 for the treated range. The noise treatments reduced the reverberation time, increased the critical distance and improved speech intelligibility in the CATF firing range.

8:30

5aAA3. Architectural acoustics in educational facilities: An empirical study on university classrooms in Egypt. Hadia S. Awad (Architecture, Alexandria University, 131 Ahmed Shawky St., Sidi Gaber, Alexandria 21529, Egypt, hadyaawad@pua.edu.eg), Hania H. Farag (Electrical Engineering, Alexandria University, Alexandria, Alexandria, Egypt), Dina S. Taha, and Mohamed A. Hanafi (Architecture, Alexandria University, Alexandria, Alexandria, Egypt)

There is a lack of acoustic performance standards for educational facilities in Egypt, especially university classrooms, resulting in a very bad acoustical quality. There is also a lack in the scientific research in that field of study in Egypt, although this type of research has been held in many other countries. This paper aims at studying the acoustic performance standards in different countries such as (ANSI-S12.60, 2010, BB93, 2003), and then using them to evaluate the acoustical quality in classrooms in Alexandria, including public and private, new and old constructions. The evaluation was done by three methods; B&K2250 SLM, Computer modeling using Odeon acoustic prediction software, and mathematical calculations using Sabine and Eyring formulae. The empirical study results were then compared to the recommended values by the previously mentioned standards. The study focused on sound levels, noise levels, reverberation time and Signal-to-Noise ratios. The results prove that there is a lack of acoustic design in Egypt. Most of the studied classrooms did not meet the recommended values. The main problem is using inappropriate finishing materials, which provide very high reverberation. Some simple and applicable proposals were suggested to improve the acoustical quality of classrooms. Predictions have shown a significant improvement.

8:45

5aAA4. Measurements of the acoustical properties of iron slag panels as porous media using scale models. Ho Jun Kim, Hyung Suk Jang, and Jin Yong Jeon (Architectural Engineering, Hanyang University, Seoul, Seongdong-gu 133791, Republic of Korea, jyjeon@hanyang.ac.kr)

Iron slag, by-product of smelting iron ore, has been widely recycled to road pavement material and concrete for environmental-friendly effect. In the present study, the acoustical characteristics of iron slag panels (porous media) were investigated by using scale model approach. The scale models of the iron panels were constructed by considering different sizes of slags. The measurements of absorption and scattering/diffusion coefficients using scale models were conducted in a 1:10 scale reverberation chamber. Through the measurements, the effects of diameters and thicknesses of slags on acoustical properties of iron slag panels were investigated.

9:00

5aAA5. Sound power measurements in non-ideal enclosures using acoustic energy density. Daniel R. Marquez, Scott D. Sommerfeldt, Timothy W. Leishman (Physics and Astronomy, Brigham Young University, Provo, UT 84604, marquezdanny77@gmail.com), and Jonathan Blotter (Mechanical Engineering, Brigham Young University, Provo, UT 84604)

Sound power measurements are generally made in reverberation or anechoic chambers using acoustic pressure measurements as outlined in specific ISO or other standards. Reverberation chambers are used to approximate diffuse acoustic fields wherein the sound power is directly proportional to the spatially averaged squared pressure. Anechoic chambers are utilized to create a direct field condition, wherein sound power can also be determined from sound pressure levels located on a measurement surface which envelopes the source. This paper will introduce a method that utilizes acoustic energy density to estimate the sound power produced in non-ideal enclosures when both direct and reverberant energies contribute significantly to the total acoustic field. Since the acoustic energy density in an enclosure is more spatially uniform than the acoustic pressure, this method can achieve the same accuracy in determining sound power with fewer measurement positions when spatially averaging. The results from numerical models of several rectangular rooms of varying acoustical properties will be presented and the accuracy of the method will be addressed

9:15

5aAA6. A comparison of the precedence effect with specular and diffusive reflections. Michael T. Pastore (Architectural Acoustics, Rensselaer Polytechnic Institute, 4 Irving Place, Troy, NY 12180, m.torben.pastore@gmail.com)

Despite early reflections arriving along many trajectories, humans can localize sounds based on the direction of the direct sound source. The ‘precedence effect’ describes a set of phenomena thought to be involved in this human ability to localize sound in what could be confusing reverberant environments. Acousticians often apply diffusive surfaces to remove echoes and reduce spatial variation in rooms designed for speech and music, yet the perceptual effects of these treatments are not well understood. The ability of listeners to localize sound is crucial to the success of an enclosed acoustic environment. Therefore the precedence effect under diffusive conditions bears some exploration. A psychoacoustic experiment attempts to characterize the effect of diffusion on the precedence effect. Using an acoustic pointer, lead/lag stimuli at inter-stimulus intervals ranging from 0–4 ms are used for comparing listeners’ perceived lateralization of a target stimulus in the presence of a single simulated specular or diffuse reflection. Bandpass noise bursts centered at 500 Hz, (ITD $\pm 300\ \mu s$) are used for the creation of all stimuli. Listeners’ performance is evaluated using existing cross-correlation models.

9:30

5aAA7. The effect of building reflections on the equivalent noise level (Leq) from traffic on Lake Shore Drive. Eric W. McGowan (Audio Arts & Acoustics, Columbia College Chicago, Chicago, IL 60614, eric.mcgowan@loop.colum.edu)

The objective of this study was to determine if the Leq resulting from vehicular traffic on Lake Shore Drive in Chicago was affected by reflections from buildings being present in the background. Three test locations were chosen along the same stretch of road, two with buildings in the background, one without. A noise propagation model using CadnaA was constructed in order to compare results from the test locations. Traffic noise spectra were recorded at each location for one (1) hour and post-processed using SpectraPLUS; test data were also acquired using a Type II SPL meter with Leq capabilities. The location with the most building coverage in the background yielded the highest Leq, while the location with no buildings gave the lowest level. The computer model produced results that followed the trends of the test, but the predicted values were consistently higher than the measured levels. The study analyzed if the effect of parameters such as pavement and country-specific Standards could account for the differences between measured and modeled data.

9:45

5aAA8. Abstract withdrawn

10:00–10:15 Break

10:15

5aAA9. Computer simulation of Benjamin Franklin’s acoustic experiment on George Whitefield’s oratory. Braxton B. Boren and Agnieszka Roginska (Music, New York University, New York, NY 10012, bbb259@nyu.edu)

The Anglican preacher George Whitefield was renowned for his loud voice and the huge crowds he drew during the transatlantic revivals of the 18th century. Benjamin Franklin was skeptical of the accounts of crowds of 30,000 gathering in London, and when Whitefield came to Philadelphia in 1739, Franklin performed one of the earliest recorded ‘archeoacoustic’ experiments: walking backwards down Market Street, Franklin continued listening to Whitefield speak from the old courthouse until his sermon became unintelligible. Using this maximum intelligible distance, Franklin calculated that Whitefield probably could have been heard by more than 30,000 listeners. Using Franklin’s account and period maps and prints of the colonial city, we have built a virtual CAD model of Philadelphia as it would have existed during Whitefield’s visit. This paper discusses techniques employed using geometric acoustic simulation software to approximate the loudness of Whitefield’s voice based on the STI at Franklin’s position. To determine the STI, the background noise at Franklin’s position is simulated according to his account of a noise source on Front Street. Given a specific noise source and a minimum intelligible STI, this system yields a loudness value in dB-SPL for an acoustic source at Whitefield’s position.

10:30

5aAA10. Streamlining sound power level measurements for determining the product noise rating for consumer products. Matthew A. Nobile (IBM Acoustics Lab, M/S P226, Bldg 704, Boardman Road Site, 2455 South Road, Poughkeepsie, NY 12601, nobile@us.ibm.com)

Under the assumption that consumer demand for quieter products will increase once noise level information is routinely available, a simplified product noise rating method has been proposed and described in several recent papers and forums (e.g., Inter-Noise 2011 and Inter-Noise 2012 papers by the author). This “PNR method” will provide useful noise information to the general public to help them make informed purchasing decisions. The essential elements of the new method are: (1) a product noise rating scale, (2) the Product Noise Rating (PNR), itself, (3) a range-of-levels indication, and (4) a visual icon for presenting the PNR value. But no matter how well-defined this new method is for providing noise levels to consumers, it is useless until the levels, themselves, have actually been measured for a wide range of products. Along this line, this paper will suggest the use of streamlined sound power level measurement procedures on typical consumer products that might help speed the acceptance and use of the new PNR method by manufacturers, product resellers, consumer testing organizations, and other stakeholders.

10:45

5aAA11. Acoustic design of transit stations. Alan Oldfield and Frank Babic (AECOM, 5600 Cancross Court, Suite A, Mississauga, ON L5R 3E9, Canada, alan.oldfield@aecom.com)

The acoustical environment in transit stations is being given increasing attention, particularly in national codes and client standards which now include requirements for intelligibility of voice alarm systems. This differs from the past where fewer loudspeakers with greater sound pressure output levels could be used to fulfill overall sound level criteria, with little consideration for alarm intelligibility. Room acoustics in stations is being considered using techniques previously reserved for performance venues. In addition, noise from tunnel ventilation systems for exhaust and emergency smoke extract is a growing concern that requires attention to meet strict criteria in often constrained spatial conditions and demanding environmental requirements. Incorporating acoustic treatment is often given a low priority

over considerations for cost, physical security and maintenance, not to mention the architect's aesthetic vision. This paper presents a review of the key challenges and best practices in acoustic design of transit stations.

11:00

5aAA12. Effects of short noise bursts on human performance and perception. Christopher Ainley and Lily M. Wang (Durham School of Architectural Engr. and Constr., University of Nebraska - Lincoln, 241 Peter Kiewit Institute, 1110 S. 67th Street, Omaha, NE 68182, cainley@unomaha.edu)

The goal of this research project is to better quantify human reactions to short bursts of noise, to complement research at NASA Langley Research Center on evaluating human response inside buildings to low-level sonic booms. The project involved exposing participants over 30-minute sessions to 250 ms broadband noise bursts of certain levels, presented in a controlled yet randomized fashion throughout the session, and gathering responses on human perception and performance on an arithmetic task dealing with short-term memory. While previous research has demonstrated effects of noise bursts of varying amplitudes on other types of tasks that study cognitive processing including attention and at louder levels on this arithmetic task (i.e. 100 dB peak), more information is needed to indicate at what level and to what degree such noise bursts may impact human performance and perception. Twenty-seven test subjects were tested over multiple sessions, with four different levels of the noise bursts, ranging from 47 to 77 dBA. The gathered performance and perception results will be presented, with consideration of each subject's self-reported noise sensitivity. The results will also be correlated to a number of noise metrics, including perceived level (PL) and L1-L99. [Work supported by a NASA Nebraska Space Grant.]

11:15

5aAA13. Community and individual variation in response to noise from high amplitude impulsive sounds. Edward T. Nykaza and Dan Valente (US Army Corps of Engineers, Engineer Research and Development Center, 2902 Newmark Drive, Champaign, IL 61822, edward.t.nykaza@erdc.dren.mil)

It is common for residents living on and around military installations to be exposed to a significant amount of high amplitude impulsive noise, primarily from large weaponry and other blast noise producing sources. Yet in comparison to transportation noise, there have been relatively few studies of how communities and individuals respond to this type of noise. This presentation will report the latest findings from recent human response to blast noise studies conducted at three military installations. Across all sites, blast noise has been found to be the most annoying noise source, despite the fact that a large percentage of respondents reported that their neighborhood was a good or excellent place to live. It has also been found that each community and individual has a unique tolerance to blast noise. Furthermore, individuals use a different and finite portion of the response scale, suggesting that the current methodology of fixating on the percent of the population that is highly annoyed may inadvertently be discarding useful response information. Comparisons between respondents living on- and off-post within and between study sites will be made, with special emphasis placed on differences between the community tolerance level and the community tolerance spread for each site.

11:30

5aAA14. Whoosh noise measurements from an automotive centrifugal compressor. Rick Dehner, Neil Figurella, Kevin Fogarty, and Ahmet Selamet (Mechanical Engineering, The Ohio State University, Columbus, OH 43212, dehner.10@osu.edu)

Broadband noise, accompanying the flow separation and turbulence, is studied for an automotive centrifugal compressor on a steady-flow turbocharger bench facility. When operated in the mid-flow region, the current compressor exhibited elevated broadband noise at high frequencies, which was evident both in the upstream compressor duct and external sound pressure level (SPL) measurement locations. Viewing SPL data as a function of the flow angle relative to the leading edge of the inducer blades (incidence angle) reveals a relationship which is nearly independent of the rotational speed. As the incidence angle is increased, broadband noise levels first go up gradually, nearly level off, and then decrease sharply at a critical incidence angle.

11:45

5aAA15. A study of background noise variability in acoustical measurement laboratories. Seth Bard (IBM Hudson Valley Acoustics Lab, M/S P226, Bldg 704, Boardman Road Site, 2455 South Road, Poughkeepsie, NY 12601, sethbard@us.ibm.com)

Presently the background noise level is included the background correction procedures in the noise emissions standards, ISO 3741, ISO 3744, ISO 3745, and ISO 11201, but the variability of the background noise is not assessed in the corrections. This paper continues the work presented in a previous paper [Proc. Inter-Noise 2012, Paper 1045] by identifying the standard deviations of the background noise sound pressure levels in multiple acoustical test laboratories for the purpose of defining the measurement capability. The foundational paper proposes using both the variability of the background noise and the difference between the source-plus-background measurement and the background measurement to calculate the background noise correction. The data obtained in the background noise study are converted into sound pressure level minima for a source measurement conforming to the requirements of the new proposal in the aforementioned foundational paper. These results are then compared with the smallest source measurements conforming to the current noise emission standards.

Session 5aSC

Speech Communication: Speech Production II: Anatomy, Models, and Methods (Poster Session)

Jody E. Kreiman, Chair
UCLA, Los Angeles, CA 90403

Contributed Papers

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

5aSC1. Progression of age-related voice instabilities in a non-pathological voice. Eric J. Hunter and Ingo R. Titze (National Center for Voice and Speech, University of Utah, Salt Lake City, UT 84101, eric.hunter@utah.edu)

Age-related structural and functional changes to the aerodigestive tract can affect breathing, swallowing, and voice. Not only can these changes can shape an individual's quality of life, they can, ultimately, be life-threatening. Looking at the voice specifically, changes to the subglottal and supraglottal airways influence vocal fold vibration by producing major bifurcations in the vibration regime. These bifurcations are evidenced in the voice by aphonic segments, subharmonic or side-band frequencies, frequency jumps, and chaotic vibration. The increased occurrence of these bifurcations can, in turn, may indicate age related changes in the vocal folds. The current study examines age-related changes in voice production in two individuals spanning 48–98 y/o and 52–90 y/o. Previous studies revealed changes in breath rate and pitch begin between the ages of 68–74 y/o, indicating a fundamental change in the body's maintenance of the speech mechanism. Voice breaks and bobbles are shown to correspond with this change. These voice breaks, along with the increased voice pitch previously reported, may indicate an interaction of the subglottal and supraglottal airway and an increased weakness of the vocal folds. Weakened vocal folds could also indicate compromised swallowing and airway protection mechanisms.

5aSC2. Spectrographic analysis as an indicator of perceived progress during speech therapy. Kathleen Siren (Speech-Language Pathology/Audiology, Loyola University Maryland, 4501 North Charles St., Baltimore, MD 21210, ksiren@loyola.edu) and Shannon Katz (Loyola Univ., Baltimore, MD)

Despite a respectable data set in speech literature documenting acoustic features of disordered speech, there is a notable scarcity of investigations documenting use of acoustic analysis during speech therapy. In 1999, Kent, Weismer, Kent, Vorperian and Duffy predicted that "joint perceptual-acoustic analysis" would soon be common in clinics due to the lower cost and ease of use of acoustic analysis procedures. Given the paucity of published data, such joint analyses are either still not occurring or are limited to use in clinical settings where information is not shared. Although some clinicians are beginning to use spectrograms as visual biofeedback during speech therapy for children, few studies have documented the effectiveness of this clinical use. Additionally, no studies have attempted to assess the relationship between spectrographic change and change in clinician perception of sound production. This investigation will assess the utility of spectrographic analysis in a clinical setting with children as they progress through speech therapy for misarticulation of /s/ by measuring the acoustic changes occurring over time as sound production matures, and providing additional information about the relationship between acoustic features and perception of accurate sound production.

5aSC3. Variance in tongue motion patterns during the production of /s/. Cindy Ding, Jonghye Woo (Neural and Pain Science, University of Maryland School of Dentistry, Baltimore, MD 21201), Hegang Chen (Epidemiology, University of Maryland School of Medicine, Baltimore, MD), and Maureen Stone (Neural and Pain Science, University of Maryland School of Dentistry, 650 W. Baltimore St Rm 8207, Baltimore, MD 21201, mstone@umaryland.edu)

One of the issues in speech motor control is the nature of variance that occurs in the production of a single speech task spoken by multiple speakers. Do they all essentially use the same gesture modified by fine tuning to adjust to their own anatomy, dialect, etc, or are there quite different, motor equivalent, ways to produce the same speech sound. Production of /s/ in American English is known to be produced using two methods: apical or laminal. Apical /s/ primarily elevates the tongue tip, while laminal /s/ utilizes the tip and blade. Both gestures are found frequently in normal speakers. The present study uses principal components analysis of midsagittal velocity fields to identify the patterns of variance in the internal tongue motion patterns of 10 normal speakers. Palate height will also be examined, as preliminary evidence points to low-palate speakers having a preference for apical /s/, while high-palate speakers use either. Tagged-MRI was used to record 'a geese', and the motion between /g/ and /s/ studied for amount and directions of variance. The goal is to identify stable features of the motion and the effects of /s/ type and palate height.

5aSC4. Single motor unit activity in the genioglossus muscle during vowel articulation. Amy LaCross, Sayoko Takano, Ian J. Kidder (Department of Physiology, University of Arizona, Tucson, AZ 85701), Peter J. Watson (Speech, Language, Hearing Sciences, University of Minnesota, Minneapolis, MN), and Elizabeth Fiona Bailey (Department of Physiology, University of Arizona, 1501 N. Campbell, Rm. 4104, PO Box 245051, Tucson, AZ 85701, ebailey@email.arizona.edu)

There is an abundance of previous electromyographic (EMG) research conducted in the human tongue that has as its focus the extrinsic tongue muscle, genioglossus (GG) and its role in preserving upper airway patency for purposes of gas exchange. Comparatively few studies have documented GG EMG activities in the performance of volitional tasks such as speech production (Honda 1992; Honda et al. 1992; Kusakawa et al 1993). Here we report on de novo efforts to characterize GG single motor unit (SMU) activities in the context of volitional i.e., speech and automatic i.e., central pattern generator driven respiratory activities. Using tungsten microelectrodes inserted into the belly of the GG in 5 healthy adults, we recorded SMU activity during static articulation of the vowels [a] and [ae] in three conditions: phonated, whispered, and articulation using an electrolarynx. Preliminary findings provide evidence of recruitment and de-recruitment of GG motor units coincident with the onset and offset of vowel articulation in phonated and whispered, but not electrolarynx conditions. Furthermore, fluctuations in GG MU firing rates mirror intensity variations within phonated and whispered utterances. These findings provide much-needed new insights into the

differential modulation of lingual motor unit activities for purposes of speech production versus breathing.

5aSC5. Numerical investigation of the influence of thyroarytenoid and cricothyroid muscle contraction on the geometry and biomechanical properties of the vocal folds. Jun Yin and Zhaoyan Zhang (UCLA School of Medicine, 1000 Veteran Ave, 31-24 Rehab Center, Los Angeles, CA 90095, zyzhang@ucla.edu)

It is generally accepted that different voice types can be produced for different stiffness and geometry conditions of the vocal folds. However, it remains unclear how the stiffness and geometry of the vocal folds are regulated through laryngeal muscle activation during phonation. A better understanding of such muscular mechanisms of regulating vocal fold properties would provide important insights into the process of physiological control of phonation. In this study, the influence of the activation of the thyroarytenoid (TA) and cricothyroid (CT) muscles on vocal fold geometry and stress distribution within the vocal fold was investigated in a three-dimensional body-cover continuum model of the vocal folds. The results showed that different combinations of the TA and CT activation levels led to different body-cover stress distributions within the vocal fold. Contraction of the TA muscle also caused the vocal fold to bulge towards the glottal midline and created a medial compression force. The results also showed that, in some conditions, coordination of different laryngeal muscles is required to produce an optimal effect on vocal fold geometry or stiffness. [Work supported by NIH.]

5aSC6. On parameterizing glottal area waveforms from high-speed images. Gang Chen (Department of Electrical Engineering, University of California, Los Angeles (UCLA), Los Angeles, CA 90095-1594, gangchen@ee.ucla.edu), Jody Kreiman, Bruce R. Gerratt (Division of Head and Neck Surgery, School of Medicine, University of California, Los Angeles, CA), Yen-Liang Shue (Dolby Australia, McMahons Point, NSW, Australia), and Abeer Alwan (Department of Electrical Engineering, University of California, Los Angeles, CA)

Because voice signals result from vocal fold vibration, perceptually-meaningful vibratory measures should quantify those aspects of vibration that correspond to differences in voice quality. In this study, glottal area waveforms were calculated from high-speed images of the vocal folds. Principal component analysis was applied to these waveforms to investigate the factors that vary with voice quality. Results showed that the first two principal components were significantly ($p < 0.01$) associated with the open quotient and the ratio of alternating-current to direct-current components. However, these conventional source measures, which are based on glottal flow, do not fully characterize observed variations in glottal area pulse shape across different glottal configurations, especially with respect to patterns of glottal closure that may be perceptually important. A source measure, the Source Dynamic Index (SDI), is proposed to characterize glottal area waveform variation for both complete and incomplete glottal closures. Analyses of "glide" phonations in which quality varied continuously from breathy to pressed showed that the SDI is able to characterize the corresponding continuum of glottal area waveform variation, regardless of the presence or absence of glottal gaps. [Work supported in part by NSF and NIH.]

5aSC7. Incorporating cepstral peak prominence as an acoustic method for assessing variation in voice quality. Katherine L. McDonald and Erik R. Thomas (English, North Carolina State University, Raleigh, NC 27695, erthomas@ncsu.edu)

Cepstral Peak Prominence (CPP) has been proposed as a means of assessing breathiness from recordings (Hillenbrand et al., J. Speech and Hearing Res. 37(1994):769-78). CPP involves computing the deviation of the cepstral peak in a smoothed cepstrum from a regression function for the smoothed cepstrum. It has not been tested extensively, though the few evaluative studies, involving comparisons against auditory judgments of voice quality, have yielded positive results. Previous evaluations have not assessed the use of CPP for comparisons of inter-group or intra-individual variation. A reliable way to gauge breathiness is useful not only for therapeutic purposes, but also for informing researchers about a speaker's repertoire and identity construction. This study examines the efficacy of CPP for analyzing social variables at the individual and group levels, using African American and European American subjects producing read and spontaneous speech under controlled conditions. Results show intra-speaker consistency across speaking tasks, but no consistent inter-ethnic differences. These findings suggest that some aspects of phonation may be more important for individual identity than for group identity.

5aSC8. Perceptual evaluation of voicing source models. Jody E. Kreiman, Bruce R. Gerratt (Head and Neck Surgery, University of California, Los Angeles, 31-24 Rehab Center, 1000 Veteran Avenue, Los Angeles, CA 90403, jkreiman@ucla.edu), Gang Chen (Electrical Engineering, University of California, Los Angeles, CA), Mark Garellek (Linguistics, University of California, Los Angeles, CA), and Abeer Alwan (Electrical Engineering, University of California, Los Angeles, CA)

Many models of the glottal source have been proposed, but none has been systematically validated perceptually, so that it is unclear whether deviations from perfect fit have perceptual importance. If model fit fails in ways that have no perceptual significance, such "errors" can be ignored, but poor fit with respect to perceptually-important features has both theoretical and practical importance. To address this issue, we fit 6 different source models to 40 natural voice sources, and then evaluated fit with respect to time-domain landmarks on the source waveforms and details of the harmonic voice source spectrum. We also generated synthetic copies of the voices using each modeled source pulse, with all other synthesizer parameters held constant, and then conducted a visual sort-and-rate task in which listeners assessed the extent of perceived match between the original natural voice samples and each copy. Discussion will focus on the specific strengths and weaknesses of each modeling approach for characterizing differences in vocal quality. [Work supported by NIH/NIDCD grant DC01797 and NSF grant IIS-1018863.]

5aSC9. Improving vocal tract reconstruction and modeling through super-resolution volume reconstruction technique. Jonghye Woo (Departments of Neural and Pain Sciences and Orthodontics, University of Maryland Dental School, Baltimore, MD), Xinhui Zhou (Department of Electrical and Computer Engineering, University of Maryland, College Park, MD 20740, zxinhui2001@gmail.com), Maureen Stone (Departments of Neural and Pain Sciences and Orthodontics, University of Maryland Dental School, Baltimore, MD), Jerry L. Prince (Department of Electrical and Computer Engineering, Johns Hopkins University, Baltimore, MD), and Carol Y. Espy-Wilson (Department of Electrical and Computer Engineering, University of Maryland, College Park, College Park, MD)

Magnetic resonance imaging has been widely used in speech production for vocal tract reconstruction and modeling. In order to observe detailed structures in the vocal tract, three orthogonal image stacks (sagittal, coronal, and axial) are usually acquired. Due to many constraints, each stack typically has an in-plane resolution which is much better than the out-of-plane resolution. Usually vocal tract modeling is based on just one of these three stacks. As a result, additional useful information revealed by the other two datasets is excluded in the vocal tract model. This study is to improve the vocal tract reconstruction and modeling by integrating information from all of the three stacks. To do so, a super-resolution reconstruction method recently developed to generate an isotropic image volume is used to integrate the three orthogonal stacks. Based on the ATR MRI database of vowel production, vocal tract models from MR images in high resolution, low resolution (simulated through downsampling), and super-resolution were built respectively and compared. The improvement in vocal tract modeling due to the super-resolution technique will be demonstrated on five vowels in terms of visualization and acoustic responses. [This research was supported by NIH R01 CA133015.]

5aSC10. Intraglottal velocity and displacement measurements in an excised larynx. Liran Oren, Ephraim Gutmark (Aerospace Engineering and Engineering Mechanics, University of Cincinnati, PO Box 670528, Cincinnati, OH 45267, orenl@mail.uc.edu), and Sid Khosla (Otolaryngology - Head and Neck Surgery, University of Cincinnati, Cincinnati, OH)

A major assumption of previously published PIV measurements in excised larynges, is that the vortices seen directly above the glottal exit during closing are due to the flow separation vortices (FSV) that formed in the

glottis. This assumption needed experimental validation. In addition, the pressures associated with the vortices above the vocal folds may be different than the pressures generated by the intraglottal FSV. Previous studies also relied on images of the glottal opening taken from above of the folds to evaluate the displacement of the folds. This method cannot separate the dynamic of the folds that occurs along their superior and inferior edges. The current study uses a major modification of the PIV techniques previously described to simultaneously measure intraglottal velocity fields and intraglottal geometry. The intraglottal pressure distribution is computed from the velocity measurements by solving the pressure Poisson equation. The results show that strong negative pressure is formed towards the superior edge of the folds. This negative pressure can produce additional force during closing. The displacement of the folds, during closing, is also extracted from the PIV images and it shows that the acceleration of the superior edge is consistently higher than the inferior edge.

5aSC11. Intraglottal pressures related to glottal and laryngeal asymmetries. Ronald Scherer (Communication Sciences and Disorders, Bowling Green State University, 200 Health Center, Bowling Green, OH 43403, ronalds@bgsu.edu)

The air pressures within the glottis during phonation may be highly dependent upon the symmetry of the glottis and other laryngeal structures. Physical laryngeal models M5 and M6 have been used to explore the slanted "oblique" glottis with different included and oblique angles, the change to three dimensions, and the presence of the arytenoid cartilages. Results suggest that (1) the greater the oblique angle, the higher the glottal entrance pressures compared to the symmetric glottis, (2) the larger the intraglottal angle, the more different the pressures are on the two glottal sides, (3) pressures are higher on the convergent side than on the divergent side of the glottis, (4) intraglottal pressure depends on which side of the glottis the flow exits, with the Flow Wall side having lower pressures, (5) for large glottal diameters, all intraglottal pressures may be lower on the Flow Wall side, (6) for special cases of high transglottal pressure, the pressure near glottal exit may be lower than at entrance (divergent glottis), (7) a symmetric but 3-dimensional glottis has consistent but minor pressure changes in the anterior-posterior direction, and (8) the presence of the arytenoid cartilages has minor effects on the intraglottal pressures. [Support from NIH.]

5aSC12. An articulatory and acoustic study of fricative consonants /s/ and /sh/ in normal and post-glossectomy speakers. Xinhui Zhou (Department of Electrical and Computer Engineering, University of Maryland, College Park, MD 20740, zxinhui2001@gmail.com), Woo Jonghye, Maureen Stone (Departments of Neural and Pain Sciences and Orthodontics, University of Maryland Dental School, Baltimore, MD), and Carol Y. Espy-Wilson (Department of Electrical and Computer Engineering, University of Maryland, College Park, MD)

Glossectomy is a surgical procedure to remove the cancerous tumor of the tongue. After the glossectomy, the tongue is sutured closed or a flap is inserted to reconstruct the tongue volume. As a result, the properties of the tongue are more or less affected by the surgery. The changes in the tongue properties may also affect the speech production abilities of the post-glossectomy speaker. This study examined the production of the fricative consonants /s/ and /sh/ in normal and post-glossectomy speakers. The data analyzed consisted of audio and magnetic resonance images from dozens of normal and glossectomy speakers. An acoustic analysis showed that the average centers of gravity of /s/ and /sh/ in glossectomy speakers are significantly lower than in normals. This difference may be explained by a more posterior constriction in glossectomees due to the surgery. Examination of the tongue shapes in midsagittal MR images showed that they tend to have more laminal /s/ than apical /s/. 3-D vocal tracts of /s/ and /sh/ were reconstructed for three glossectomy speakers whose /s/ and /sh/ cannot easily be discriminated in listening tests. Details of the 3-D vocal tract shapes, along with their acoustic implications, will be discussed for the glossectomy and normal speakers. [This research was supported by NIH R01 CA133015.]

5aSC13. Acoustic discrimination of Parkinsonian speech using cepstral measures of articulation. Mark D. Skowronski, Rahul Shrivastav (Communicative Sciences and Disorders, Michigan State University, Lansing, MI 48824, markskow@hotmail.com), James Harnsberger (Linguistics, University of Florida, Gainesville, FL), Supraja Anand, and Jay Rosenbek (Speech, Language, and Hearing Sciences, University of Florida, Gainesville, FL)

The effects of idiopathic Parkinson's disease (IPD) on speech include articulatory imprecision. We quantified articulation rate and range acoustically using cepstral coefficients to represent vocal tract settings. Cepstral coefficients were extracted from 10 sentences spoken by 76 talkers, half of which were diagnosed with IPD. Articulation range was estimated for each sentence as the sum across cepstral coefficients of the standard deviation of each coefficient, and articulation rate was estimated using the same procedure, replacing cepstral coefficients with delta coefficients. The mean $\pm$ standard deviation (N = 380 sentences) for the articulation measures of range (7.95 ± 0.50 vs. 6.66 ± 0.53) and rate (5.64 ± 0.56 vs. 4.40 ± 0.46) were significantly different (t-test, $p < 0.001$) for normal vs. IPD speech, respectively. In a leave-one-talker-out classification experiment, range accuracy was 90.1%, rate accuracy was 88.8%, and accuracy was 92.9% using a combined model of articulation range and rate. The strengths of the articulation measures include 1) sensitivity to IPD speech, 2) reliance on cepstral coefficients which have been used for over 30 years to represent speech, 3) no segmentation requirements, 4) low sensitivity to speech material, and 5) effective with only 2 seconds of speech.

5aSC14. Second-formant locus patterns in dysarthric speech. Heejin Kim and Mark Hasegawa-Johnson (Beckman Institute, University of Illinois at Urbana-Champaign, Urbana, IL 61801, hkim17@illinois.edu)

Second-formant (F2) locus equations represent a linear relationship between F2 measured at the vowel onset following stop release and F2 measured at the vowel midpoint in a consonant-vowel (CV) sequence. Prior research has used the slope and intercept of locus equations as indices to coarticulation degree and the consonant's place of articulation. This presentation addresses coarticulation degree and place of articulation contrasts in dysarthric speech, by comparing locus equation measures for speakers with cerebral palsy and control speakers. Locus equation data are extracted from the Universal Access Speech (Kim et al. 2008). The data consist of CV sequences with labial, alveolar, velar stops produced in the context of various vowels that differ in backness and thus in F2. Results show that for alveolars and labials, slopes are less steep and intercepts are higher in dysarthric speech compared to normal speech, indicating a reduced degree of coarticulation in CV transitions, while for front and back velars, the opposite pattern is observed. In addition, a second-order locus equation analysis shows a reduced separation especially between alveolars and front velars in dysarthric speech. Results will be discussed in relation to the horizontal tongue body positions in CV transitions in dysarthric speech.

5aSC15. Temporal structure in the speech of persons with Dementia of the Alzheimer's Type. Linda Carozza, Pamela Cantor (Communication Sciences & Disorders, St. John's University, Queens, NY), and Fredericka Bell-Berti (Communication Sciences & Disorders, St. John's University, 8000 Utopia Parkway, Queens, NY 11439, bellf@stjohns.edu)

Although cognitive and language processes in dementia have been studied extensively, the question of motor speech degeneration in the course of dementing illness is a relatively unexplored area. The potential for early dissociation of motor functions of language at the level of speech production has not been explored; an interaction between motor speech and language production and perception changes should inform our understanding of the deterioration in dementia. In previous reports on two persons with DAT, we have shown inconsistent final lengthening and effects of syllable-final consonant voicing on vowel duration for one of the two speakers. We recorded one of the speakers again, and his speech was markedly slower. In this report, we expand our analysis to include three additional persons with mild-to-moderate DAT, from whom a series of 4-word phrases containing a target word occurring in phrase-medial or phrase final position was elicited. We will present the results of our analysis of final lengthening, compensatory shortening, and the effects of final consonant voicing on vowel duration.

5aSC16. Articulatory-to-acoustic relations in response to speaking rate modulation in talkers with amyotrophic lateral sclerosis. Antje Mefferd and Stephanie Entz (Wichita State University, Wichita, KS 67260, antje.mefferd@wichita.edu)

The purpose of this study was twofold. One goal was to determine the effects of speaking rate modulation on tongue kinematic and vowel acoustic distinctiveness in talkers with amyotrophic lateral sclerosis (ALS). Another goal was to determine the strength of articulatory-to-acoustic relations in response to speaking rate modulations in talkers with ALS. Six talkers with mild ALS and six healthy controls repeated "See a kite again" at their habitual rate, at a fast rate and a slow rate. The posterior tongue motion was captured simultaneously with the acoustic signal using a 3D electromagnetic articulograph (AG500). To determine kinematic and acoustic distinctiveness maximum tongue excursions and F1/F2 vowel space distance were calculated for the diphthong "ai" in "kite." Preliminary findings showed a greater effect of rate modulation on acoustic distinctiveness than on articulatory distinctiveness for both groups of speakers. The predictability of acoustic distinctiveness based on articulatory distinctiveness varied greatly amongst talkers of both groups. Findings provide empirical evidence of quantal relations between incremental changes of vocal tract configuration and vowel acoustics. Further, findings yield important clinical implications to improve intelligibility and potential explanations for speaking rate declines in talkers with ALS.

5aSC17. Co-registration of articulographic and real-time magnetic resonance imaging data for multimodal analysis of rapid speech. Jangwon Kim (Electrical Engineering, University of Southern California, 3740 McClintock Avenue, Los Angeles, CA 90089, jangwon@usc.edu), Adam Lammert (Computer Science, University of Southern California, Los Angeles, CA), Michael Proctor, and Shrikanth Narayanan (Electrical Engineering, University of Southern California, Los Angeles, CA)

We propose a method for co-registrating speech articulatory/acoustic data from two modalities that provide complementary advantages. Electromagnetic Articulography (EMA) provides high temporal resolution (100 samples/second in WAVE system) and flesh-point tracking, while real-time Magnetic Resonance Imaging, rtMRI, (23 frames/second) offers a complete midsagittal view of the vocal tract, including articulated structures and the articulatory environment. Co-registration was achieved through iterative alignment in the acoustic and articulatory domains. Acoustic signals were aligned temporally using Dynamic Time Warping, while articulatory signals were aligned variously by minimization of mean total error between articulatometry data and estimated corresponding flesh points and by using mutual information derived from articulatory parameters for each sentence. We demonstrate our method on a subset of the TIMIT corpus elicited from a male and a female speaker of American English, and illustrate the benefits of co-registered multi-modal data in the study of liquid and fricative consonant production in rapid speech. [Supported by NIH and NSF grants.]

5aSC18. Discriminating vocal tremor source from amplitude envelope modulations. Kathy M. Carbonell, Brad Story, Rosemary Lester, and Andrew J. Lotto (Speech, Language & Hearing Sciences, University of Arizona, Tucson, AZ 85721, kathyc@email.arizona.edu)

Vocal tremor can have a variety of physiological sources. For example, tremors can result from involuntary oscillation of respiratory muscles (respiratory tremor), or of the muscles responsible for vocal fold adduction (adductory tremor) or lengthening (f0 tremor). While the sources of vocal tremor are distinct, they are notoriously difficult to categorize both perceptually and acoustically. In order to develop acoustic measures that can potentially distinguish sources of tremor, speech samples were synthesized using a kinematic model of the vocal folds attached to a model of the vocal tract and trachea [Titze, JASA, 75, 570-580; Story, 2005, JASA, 117, 3231-3254]. Tremors were created by modulating parameters of the vocal fold model corresponding to the three types mentioned above. The acoustic measures were related to temporal regularities in the amplitude envelope computed across the entire signal and select frequency bands. These measures could reliably categorize the samples by tremor source (as determined from a discriminant function analysis) even when compound tremors (created from more than one source) were included. These results supply initial support for an acoustic based approach to diagnosing tremor source and further evidence for the rich information about talker characteristics present in the temporal structure of the amplitude envelope.

5aSC19. Principal components analysis comparison of normal and glossectomy movement patterns in multiple tasks. Caitlin R. Gallagher, Jonghye Woo (Neural and Pain Science, University of Maryland School of Dentistry, Baltimore, MD), Hegang Chen (Epidemiology, University of Maryland School of Medicine, Baltimore, MD), and Maureen Stone (Neural and Pain Science, University of Maryland School of Dentistry, 650 W. Baltimore St Rm 8207, Baltimore, MD 21201, mstone@umaryland.edu)

Recent estimates suggest that 34,000 people are diagnosed with oral cancer each year. The lateral border of the tongue is one of the most common sites for lingual cancer and surgery resections both muscles and nerves leading to the tongue tip. One sound that is typically impaired is /s/ as it requires precise tongue shape, location and palatal contact, and small errors are acoustically salient. This study uses Principal Components Analysis (PCA) to compare motion patterns of the internal tongue during the word 'geese'. The study will compare 4 subjects: one glossectomy patient and a matched control who produce an apical /s/, and another pair that produces a laminal /s/. A PCA will be run for all subjects and the principal patterns of variance will be determined. These patterns will be used to identify stable features of the motion and variations due to /s/-type and patient vs control. The complexity of each subject's motion pattern will be studied to determine whether the patients have more variability in their motion due to strategies of motor adaptation, or less variability due to scarring and increased rigidity.

5aSC20. The role of respiratory/phonatory training and acoustic analysis in normalizing speaking rate in a case of scanning speech. Emily Q. Wang, Samantha A. Sanders (Department of Communication Disorders and Sciences, Rush University Medical Center, Chicago, IL 60523, emily_wang@rush.edu), and Hanjun Liu (Department of Rehabilitation Medicine, The First Affiliated Hospital, Sun Yat-sen University, Guangzhou, Guang Dong, China)

In this single case study, we report the role of respiratory/phonatory training in normalizing speaking rate in a patient with significant ataxic dysarthria, specifically, scanning speech, two years post a closed head injury from an MVA. The patient passed bilateral hearing screening, and scored 30/30 on the Mini-Mental State Exam, AQ of 97.2 on the Western Aphasia Battery-Revised and 58/60 on the Boston Naming Test. The initial speech analysis revealed the following: perceptually, the speech was slow with equal and excess stress and significant phonatory-prosodic insufficiency. Acoustically, the syllable duration was increased and equalized with very little variation in fundamental frequency or intensity. The phonatory-prosodic insufficiency was identified as the underlying deficit and targeted, which resulted in significant improvement in both overall speech intelligibility and naturalness of speech in seven weeks. Both self-rating and acoustic analyses were used throughout as means of feedback which contributed to the success of the treatment. The final acoustic analysis indicated normal syllable duration and speech prosody in spontaneous speech. The pre- and post-treatment audio clips will be used to accompany the acoustic analysis. The role of acoustic analysis in treatment of phonatory-prosodic insufficiency and its implication in dysarthria management in general will be discussed.

5aSC21. The effect of portable DAF usage in daily life on the speech intelligibility of dysarthrias. Eiji Shimura (Department of Speech, Language and Hearing Sciences, Niigata University of Health and Welfare, 1398, Shimami-cho, Kita-Ku, Niigata 950-3198, Japan, shimura@nuhw.ac.jp) and Kazuhiko Kakehi (Institute for Advanced Studies in Artificial Intelligence, Chukyo University, Toyota, Aichi, Japan)

Dysarthria is a neurologic motor speech impairment due to a pathological change of nerve and muscle system. Several rehabilitation methods of speaking rate control have been widely used to improve speech intelligibility for mild dysarthria. Nevertheless, speech intelligibility of the dysarthric patients show almost no improvement outside of a clinic. Delayed auditory feedback(-DAF)is one of the speaking rate control methods. Recently a small portable DAF device has been developed, which enables dysarthric patients to use it in their daily life. In this study, wearing a portable DAF, two dysarthric patients conducted a 20-minute practice per day for three months. In the practice, they instructed to prolong vowel length in DAF usage. Intelligibility tests for single-word utterance and for reading aloud a long sentence were conducted before and after the practices. As a result, the single-word intelligibility score increased from 84.0% to 90.6% in case1, and from 68.3% to 89.7% in case2.

Additionally, vowel space in the F1-F2 space in case2 enlarged. Both the performance of long sentence reading and free conversation will be presented in the meeting. These results suggest that a portable DAF effectiveness in daily use. This work was supported by JSPS KAKENHI 10424889.

5aSC22. Automated prosodic labeling using soft computing. Nicholas Bacuez (French and Italian, University of Texas, Austin, TX 78712, nicholasbacuez@utexas.edu)

Automated prosodic modeling/labeling usually relies on complex algorithms. However, as phonological research suggests, human beings do not process intonation in terms of algorithms but in terms of relative oppositions, linguistically encoded in words such as 'higher', 'lower', 'longer', 'shorter', and reflected by the notation: H, L, H*, L% … I have conceived an automated procedure -the 4 layer structure- based on linguistic human reasoning that locates the tones of each individual sentence in a corpus. The original time/f0 information (1st level) is scaled onto a 100 by 100 cartesian plane preserving relative oppositions of time and f0 (2nd level). At the pretonal-anchoring level (3rd level), syllables are divided into frames within which the highest and lowest points are marked as pretones; the contour is turned into a string of pretones (4 pretones per syllable) anchored to their position in the sentence. Finally, a multi-pass procedure (4th level) iteratively locates highest and lowest pretones forming larger movements until the entire sentence has been labeled. The procedure uses minimal logical and mathematical tools to quantify the difference between pretones ('higher' or 'lower') and the span of movements (consecutive 'lower' or 'higher'). It was successfully applied to various corpora of French.

5aSC23. Effects of preceding consonant features on /ð/ assimilation. Christina Schwaller (English: Linguistics, North Carolina State University, Raleigh, NC 27695, cschwaller@gmail.com)

Sociolinguists have occasionally mentioned regressive assimilation of the voiced interdental fricative /ð/ to a preceding consonant as a feature of some dialects. However, little research has been published on the topic. This study examines effects of preceding consonant features on /ð/ assimilation, using interviews with residents of Robeson County, NC. Tokens of phonemic /ð/ were examined in spectrogram form to identify which ones were assimilated. Impressionistic judgment was used in conjunction with the spectrograms. Discontinuities within consonant clusters served as evidence of unassimilated tokens. For manner of articulation, features such as stop bursts and frication noise were identified. When necessary, measurements of F2 and F3 at adjacent vowel transitions and midpoints were compared to determine place of articulation. Place, manner, voicing, and nasality of preceding consonants were then recorded. A logistic regression indicated the significance of each variable. This showed that preceding consonants affect /ð/ assimilation. Preceding alveolars are most likely to result in assimilation, which was shown to be significant. The difference in manner between fricatives and stops is also significant, as /ð/ is more likely to assimilate to a fricative. Nasality is highly significant, with /ð/ more likely to assimilate to a nasal than to an oral consonant.

5aSC24. Locus equations as measures of consonantal variation in Wisconsin English: Revisiting the vowel-to-consonant transition. Michael J. Fox (English, North Carolina State University, Rayleigh, NC 27695)

Extant literature on vowel-to-consonant (VC) locus equations (defined as a regression line fit to F2 transition measurements of one consonant paired with many vowels) suggests that they lack the high linearity of the corresponding consonant-to-vowel ones (CV) [Sussman et al. 1997]. Measurements of the second formant taken at the vocalic midpoint and last glottal pulse yielded four VC locus equations from 31 respondents from west-central Wisconsin for the syllable-final consonants /d, t, g, k/ (n = 124). Previous work with WI English indicates the existence of a differential in locus equations for the voiced vs. voiceless velar stops [Purnell 2008]. Apical stops are included for comparison across place of articulation (i.e. apical to velar) with no differential expected. High levels of linearity were found for VC locus equations for all the consonants examined. Fitting second-order locus equations [Chennoukh, et al. 1995, 1997] to the coefficients revealed a differential between /g/ and /k/, but not /d/ and /t/. Moreover, discriminant analyses yielded higher classification rates for locus equations than for token level data. These results run counter to previous characterizations of VC locus equations and suggest the potential for the use of locus equations as measures of dialect-specific coarticulation.

5aSC25. Sensitivity of acoustic parameters of /s/ in adolescents. Christine H. Shadle (Haskins Laboratories, 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu), Laura L. Koenig (Comm. Sci. & Disorders, Long Island University, Brooklyn, NY), and Jonathan Preston (Comm. Sciences, Southern CT State Univ, New Haven, CT)

Acoustic parameters that are closely related to the source and filter properties of fricatives have been shown in previous work to be useful for comparing fricative spectra, as for instance in analyzing a database of /s/ productions by typical and misarticulating children aged 10 to 15. Those results were used to study the sensitivity of different parameters designed to measure the same underlying property. One parameter, the frequency of the main front-cavity resonance, is used to compute two others that measure the effect of labialized contexts, and the degree of sibilance; therefore, three methods of finding that resonance automatically were compared. Gender differences were found in the main resonance frequency and in measures of noise source growth during the fricative, so the use of two sets of heuristically-defined frequency bands were compared, as were spectral slope vs. sound level differences. The back-cavity resonances at low frequencies tend to be more prominent in typical children's /s/ spectra than adults', and still more so in certain types of misarticulation. Different measures of that prominence were compared. The parameters most sensitive to the difference between typical children's and adult's productions, and typical and misarticulating children's productions of /s/, will be described.

5aSC26. Segmenting American English V+/l/ and V+/r/ sequences: Methodological implications. María Riera and Joaquín Romero (Estudis Anglesos i Alemanys, Universitat Rovira i Virgili, Av. Catalunya, 35, Tarragona 43002, Spain, maria.riera@urv.cat)

This paper presents some methodological implications for the segmentation of final V+/l/ and V+/r/ sequences in American English stressed monosyllables. Between the two segments in the sequences a transitional schwa-like element characterized by variable durational and spectral values as a function of the preceding vowel, the following consonant and speaking rate can be identified. Given the dynamic nature of this element, problems related to boundary placement often arise. A segmentation method based solely on spectrographic observation and auditory corroboration has proven in previous studies to be too subjective to be reliable. A more objective method based on first derivative curve extraction, which provides us with first derivative formant traces that show peaks of formant change given by velocity maxima and minima, is more suitable for our purposes. However, the decision as to whether to choose F1, F2 or F3 traces as reference points for boundary placement, together with the presence of too many peaks in some cases and of not enough peaks in other cases, poses problems to the segmentation procedure and makes it necessary for the subjective method of segmentation based on spectrographic observation and auditory corroboration to come into play. Both methods thus complement each other.

5aSC27. Using partially separable functions to image spatiotemporal aspects of Arabic pharyngealization. Ryan Shosted (Linguistics, University of Illinois at Urbana-Champaign, 707 S Mathews Ave, Urbana, IL 61801, rshosted@illinois.edu), Maojing Fu (Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, Urbana, IL), Abbas Benmamoun (Linguistics, University of Illinois at Urbana-Champaign, Urbana, IL), Zhi-Pei Liang (Electrical and Computer Engineering, University of Illinois at Urbana-Champaign, Urbana, IL), and Bradley P. Sutton (Bioengineering, University of Illinois at Urbana-Champaign, Urbana, IL)

It has been challenging to estimate the temporal domain of pharyngealization in Arabic. Conventional MRI has limited assessment of dynamic pharyngeal shape during speech. In this study, fast spiral sequences, combined with partially separable functions, were used to achieve a relatively high spatiotemporal resolution (2.2 mm $\times$ 2.2 mm $\times$ 8.0 mm, at a frame rate of 86 fps) during dynamic speech imaging of a single midsagittal slice. One male speaker of Levantine Arabic produced pairs of words that differed minimally by one speech sound: pharyngeal fricative /ħ/ or non-pharnyngeal /b/. Each word was produced 23 times. The temporal extent of pharyngeal tissue displacement associated with /ħ/ was investigated. Sounds were segmented with reference to a simultaneous, noise-canceled acoustic recording. Spatiotemporal maps of differential pixel intensity (interpreted as tissue displacement) were generated for each segment preceding the pharyngeal /

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non-pharyngeal test segment. Average differential pixel intensity in the pharyngeal area was then sampled during these preceding segments. T-tests revealed significant differences ($p < 0.01$) up to two segments away from the pharyngeal / non-pharyngeal test segment. This technique should permit investigation of spatiotemporal aspects of pharyngealization across different varieties of Arabic, where distance and direction of pharyngealization are said to vary systematically.

5aSC28. An experimental comparison of fundamental frequency tracking algorithms. Hongbing Hu and Stephen Zahorian (Electrical and Computer Engineering, State University of New York at Binghamton, PO Box 6000, Binghamton, NY 13902, hongbingh@gmail.com)

"Yet another Algorithm for Pitch Tracking -YAAPT" was published in a 2010 JASA paper (Zahorian and Hu). Although demonstrated to provide high accuracy and noise robustness for fundamental frequency tracking for both studio quality speech and telephone speech, especially as compared to other well-known algorithms (YIN, Praat, RAPT), YAAPT has not been widely used, possibly due to the difficulty of using it and uncertainty about its effectiveness for difficult conditions. Therefore, more work has been done to improve the algorithm and especially to improve its functionality and ease of use as MATLAB functions. In the present paper, the current version of YAAPT is presented, along with clear documentation for using it, both stand alone and as a function to be called by another program. Experimentally, YAAPT is compared with YIN, Praat, RAPT, and a cepstrum method for studio bandwidth speech and telephone speech for a variety of noise conditions. Experiments are conducted with multiple databases, including American English, British English, and Mandarin Chinese. For most conditions evaluated, YAAPT gives better performance than the other fundamental frequency trackers.

FRIDAY MORNING, 26 OCTOBER 2012 MARY LOU WILLIAMS A/B, 8:30 A.M. TO 11:45 A.M.

Session 5aUW

Underwater Acoustics and Acoustical Oceanography: Boundary Interaction and Inversion

Jorge E. Quijano, Chair
School of Earth and Ocean Sciences, University of Victoria, Victoria, BC V8P 5C2, Canada

Contributed Papers

8:30

5aUW1. High-frequency acoustic backscattering from a sand sediment: Experiments and data/model comparisons. Brian T. Hefner, Anatoliy N. Ivakin, and Darrell R. Jackson (Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, WA 98105, hefner@apl.washington.edu)

In the Spring of 2012, high-frequency backscattering from a sandy sediment was measured in the Gulf of Mexico at the site of the upcoming, ONR-sponsored reverberation experiment. The measurements were made using an array of sources and receivers that collected data from 200 to 500 kHz and that could be rotated such that the incident grazing angles varied from 10 to 50 degrees. This array was used previously to measure scattering from a sand/mud sediment during the Sediment Acoustics Experiment 2004 (SAX04). To support data/model comparisons, the seabed roughness, sediment shell content, sediment sound speed, and sediment attenuation were also measured. For scattering below the critical grazing angle, sediment roughness is found to be the dominant scattering mechanism while above the critical angle, roughness scattering underpredicts the measured scattering strength. To understand the scattering strength at high grazing angles, scattering from shells and shell hash is considered. The measured scattering strengths and environmental properties at the experiment site are also compared to those made during SAX04. (Work supported by the US Office of Naval Research)

8:45

5aUW2. The effect of bottom layering on the acoustic vector field. David R. Dall'Osto and Peter H. Dahl (Mechancial Engineering and the Applied Physics Laboratory, University of Washington, Seattle, WA 98103, dallosto@u.washington.edu)

A signal reflected from a layered sea-bed contains information pertaining to the sediment properties. Typically, a signal intended to probe the sea-bed is designed to have a large bandwidth to allow for time separation of arrivals from the multiple layers. Depending on the geometry, it may impossible to avoid interference of these arrivals. The interference of these multiple arrivals does establish a pattern observable in the vector intensity. Measurements of the vertical complex acoustic intensity of a near-bottom source (~λ from the seafloor) collected off the coast of New Jersey in 2006 demonstrate the effect of a sub-bottom layer and the observable interference pattern between the first bottom reflection and the sub-bottom reflection. The spatial structure of the complex intensity can be used to infer bottom properties, which are in close agreement with a number of experimental studies at this location. The observable in the complex intensity can also be directly measured with a particle motion sensor. Parabolic equation simulations of the experimental site are used to demonstrate both the characteristic of the vector field and the sensitivity of these vector properties to changes in the sediment properties.

9:00

5aUW3. A broadband model for a range of ocean sediments. Nicholas Chotiros and Marcia J. Isakson (Applied Research Laboratories, Univ. of Texas at Austin, PO Box 8029, Austin, TX 8713-8029, chotiros@arlut.utexas.edu)

In the context of the Biot theory of sound propagation in porous media, particularly water-saturated granular media, Yamamoto and Turgut [J. Acoust. Soc. Am. 83(5), 1744–1751, May, 1988] have shown that the pore size distribution can have a profound effect on the frequency dependence of the attenuation of sound. In sandy sediments below the characteristic frequency, the attenuation is predicted to increase as the second power of frequency. In soft sediments, it is found that the rate of increase is closer to the first power. By adjusting the width of the pore size distribution, it is possible to smoothly change from the second power of frequency to the first power, in certain frequency bands. This suggests that pore size distribution may be a critical parameter in the determination of sound attenuation in the seabed. The model predictions are compared to measurements from the Shallow Water 2006 experiment as an illustration. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

9:15

5aUW4. Measurement and modeling of Scholte waves in shallow water.
Gopu R. Potty, James H. Miller (Ocean Engineering, University of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882, potty@egr.uri.edu), and Marcia Isakson (Applied Research Laboratories, The University of Texas at Austin, Austin, TX)

Shear speeds in semi-consolidated and consolidated shallow water sediments can significantly impact compressional wave attenuation and arrival times of acoustic normal modes. In addition shear properties of sediments are directly related to the strength of the sediments in geotechnical applications. All of these factors emphasize the importance of estimating shear speeds in shallow water sediments. One of the most promising approaches to estimate shear speed is to invert the shear speed profile using the dispersion of interface waves (Scholte waves). The propagation speed and attenuation of the Scholte wave are closely related to shear-wave speed and attenuation over a depth of 1–2 wavelengths into the seabed. Data from the tests conducted in Narragansett Bay and off Block Island in water depths ranging from 10 m to 25 m will be presented. Modeling of interface waves will be carried out using Finite Element Method (FEM) and a wave number integration model (OASES). Sediment properties will be inferred based on the modeling and data-model comparison. [Work supported by Office of Naval Research.]

9:30

5aUW5. Validity of first-order perturbation theory for scattering from one-dimensional and two-dimensional rough surfaces described by power-law spectra.
Bryant M. Tran (Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bmtran@utexas.edu), Sumedh Joshi (Center for Applied Mathematics, Cornell Univ., Ithaca, NY), and Marcia J. Isakson (Applied Research Laboratories, The University of Texas at Austin, Austin, TX)

First-order perturbation theory is a widely used model for estimating the backscatter of acoustic waves incident on a rough surface. The validity of perturbation theory for one-dimensional surfaces described by Gaussian spectra is well established. However, little has been done to confirm its range of validity when expanded to two-dimensional surfaces. Furthermore, the range of validity for surfaces described by power-law spectra has not been fully explored. This work seeks to benchmark first-order perturbation theory against a finite element method solution for scattering from one-dimensional and two-dimensional rough pressure-release surfaces described by power-law spectra. The relationship between ranges of validity of 1D and 2D surfaces will be considered. [Work sponsored by the Office of Naval Research, Ocean Acoustics.]

9:45

5aUW6. Broadband synthetic aperture matched field geoacoustic inversion with a single hydrophone.
Bien Aik Tan, Caglar Yardim, Peter Gerstoft, and William Hodgkiss (Marine Physical Laboratory, Scripps Institution of Oceanography, University of California San Diego, 9500 Gilman Drive, La Jolla, CA 92093-0238, btan@ucsd.edu)

Traditionally matched-field geoacoustic inversion experiments sampled the acoustic field on long arrays and require powerful transmissions in order to reduce parameter uncertainty. However, single-hydrophone based geoacoustic inversion methods exist. Practically, these methods are attractive compared to the ones using long arrays. In a single hydrophone setup, spatial diversity is traded off for frequency diversity; the source is broadband. This paper uses single-hydrophone frequency coherent matched-field inversion and exploits extended-duration coherent transmissions (multiple LFM chirps) to increase signal to noise ratio. As a result, the overall signal becomes Doppler/motion intolerant. But Doppler can be modeled by including source and hydrophone horizontal motions. To correlate well with the measured field across a receiver trajectory and to incorporate a transmission across source trajectory, Doppler in waveguide and normal mode theory are applied. The method is demonstrated with 100-900 Hz LFM SW06 data with low signal to noise ratio.

10:00–10:15 Break

10:15

5aUW7. Range and cross-range resolution from a three-dimensional linearized perturbative inversion scheme.
Christopher M. Bender, Megan S. Ballard (Appl. Res. Labs, Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, bender.christopher@gmail.com), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs, Univ. of Texas at Austin, Austin, TX)

The overall goal of this work is to develop a sparse autonomous observing system to sample the four-dimensional ocean. For this purpose, a perturbative inversion scheme [S.D. Rajan, *et. al*., J. Acoust. Soc. Am., **82**, pp. 998-1017 (1987)] is applied to estimate water-column sound-speed in all three spatial dimensions at a single "snapshot" in time. The input data to the inversion are estimates of modal travel time calculated from measurements from a distributed network of acoustic sources and receivers. Temporal variability is assessed by carrying out repeated inversions for new realizations of the input data. In initial applications of the inversion scheme, out-of-plane propagation effects were ignored and the solution was obtained by assuming straight-line paths in the horizontal plane. The effect of neglecting horizontal refraction on solution accuracy is quantified. The vertical and horizontal resolution of the solution depends on the quantity of input data and on the quantity of sources and receivers, respectively. Thus, for a particular environment, the available data may be insufficient to quantify the given variability, resulting in uncertainty in the solution. This talk explores the effects of environmental variability and spatial resolution on the uncertainty of the solution. [Work supported by ARL:UT IR&D.]

10:30

5aUW8. Reconstructing surface wave profiles from reflected acoustic pulses.
Sean Walstead (ECE, UCSD, 9500 Gilman Drive, 0407, La Jolla, CA 92093-0407, swalstead@ucsd.edu) and Grant Deane (SIO, UCSD, La Jolla, CA)

Surface wave shapes are determined by analyzing underwater reflected acoustic signals. The acoustic signals (of nominal frequency 200 kHz) are forward scattered from the underside of surface waves that are generated in a wave tank and scaled to model smooth ocean swell. An inverse processing algorithm is designed and implemented to reconstruct the surface displacement profiles of the waves over one complete period. The inverse processing uses the surface scattered pulses collected at the receiver, an initial wave profile (two are considered), and a broadband forward scattering model based on Kirchhoff's diffraction formula to iteratively adjust the surface until it is considered optimized or reconstructed. Two physical length scales over which information can be known about the surface are confirmed. An outer length scale, the Fresnel zone surrounding each specular reflection point, is the only region where optimized surfaces resulting from each initial profile converge within a resolution set by the inner length scale, a quarter-wavelength of the acoustic pulse. The statistical confidence of each optimized surface is also highest within a Fresnel zone. Future design considerations are suggested such as an array of receivers that increases the region of surface reconstruction by a factor of 2 to 3.

10:45

5aUW9. A true-depth passive fathometer.
Jorge E. Quijano, Stan E. Dosso, and Jan Dettmer (School of Earth and Ocean Sciences, University of Victoria, Victoria, BC V8P 5C2, Canada, jorgeq@uvic.ca)

This paper applies a sequential trans-dimensional (trans-D) Monte Carlo algorithm for geoacoustic inversion to bottom-loss data estimated from wind-driven ambient noise at a drifting vertical array. The approach explored in this work provides range-dependent estimates of geoacoustic parameters and true-depth layering structure of the seabed, together with corresponding uncertainties. The Bayesian inversion is applied to incoherent estimates of seabed bottom loss, computed as the array drifts along a range-dependent track. The method adopts a layered representation of the seabed, where each layer is determined by sound speed, density, attenuation, and thickness. The number of layers is also included as an unknown parameter, which allows data-driven parametrization rather than an arbitrary choice of the parametrization for the seabed model. The trans-D Bayesian inversion

method samples the joint posterior probability density of all model parameters to provide parameter estimates and uncertainties. A particle filter is used to update the estimated geoacoustic parameters from one array position to the next. The sequential inversion approach is demonstrated using data from the Boundary 2003 experiment, and compared to images of the seabed layering structure obtained by an active seismic system.

11:00

5aUW10. Sound speed measurement with direct-sequence spread spectrum signal combining time of flight and phase shift. Shen Zhao, Chunjie Qiao, Yue-ke Wang, and Zhi-gang Huang (Department of Instrumental Science and Technology, Mechatronics and Automation School, National University of Defense Technology (NUDT), Chang Sha, Hu Nan 410073, China, quickjobs@163.com)

Increasing demands for high accuracy and rapid measurement of sound speed have prompted the development of portable sound velocimeter in oceanography. Generally, sound speed can be established by CTD or TD (Time Delayed) method respectively. CTD method suffers from errors of corresponding sensors and application range of the converting equations. TD method, sing-around technique representative, is widely used in portable velocimeter. The accuracy of TD is proportional to receiving SNR and signal length. The duration of tone burst in convenient sing-around is limited by multi-echo, thus only a few valid data can be utilized. To address the problems, DSSS (Direct-Sequence-Spread-Spectrum) signal in continuous form is adopted to extract the TD directly, with which sound speed could be established related to laboratory derived equations, Del Grosso's equation representative. Based on this method, two separate transducers were deployed with constant distance. Cross-correlation between received and transmitted (orthogonal form) signals is computed. Combine Time-of-Flight and Phase-Shift, the TD can be derived by two processes, the peak detecting of cross-correlation envelope and carrier phase estimating. Theoretic analyses and experiments indicate that the accuracy with DSSS relates to the period of PN sequence, the frequency of chip, and the linearity characteristic of electronic circuits and transducers.

11:15

5aUW11. An inverse method for estimating sediment sound speed in the ocean. Tao Lin and Zoi-Heleni Michalopoulou (Department of Mathematical Sciences, New Jersey Institute of Technology, 323 ML King Blvd, Newark, NJ 07102, michalop@njit.edu)

In this work, a new inverse method for estimating sediment sound-speed profiles is investigated. Stickler and Deift derived a trace formula for recovering a seiment sound-speed profile by simply using a reflection coefficient at very low frequencies. The method may be sensitive to noise and also involves computationally cumbersome calculations. In our approach, we first design a linear approximation for the trace formula based on the Born approximation, in order to reduce the computational cost. The stability of the modified inverse algorithm is then tested with synthetic noisy data. Finally, we look into ways for relaxing the limiting assumptions of the approach. [Work supported by ONR and the NSF CSUMS program.]

11:30

5aUW12. Application of smoothing techniques to sequential geoacoustic inversion. Caglar Yardim, Peter Gerstoft, and William S. Hodgkiss (Scripps Institution of Oceanography, 9500 Gilman Dr, La Jolla, CA 92093-0238, cyardim@ucsd.edu)

Sequential Bayesian techniques such as particle filters have been successfully used to track a moving source in an unknown, complex, and evolving ocean environment. These methods treat both the source and the ocean parameters as non-stationary unknown random variables and sequentially estimate the best solution in addition to the uncertainties in the estimates. Particle filters are numerical methods called sequential Monte Carlo techniques that can operate on nonlinear systems with non-Gaussian probability density functions. Particle smoothers are a natural extension to the filters. A smoother is appropriate in applications where all data have already been observed and are readily available. Therefore, both past and future measurements can be exploited. Geoacoustic and source tracking is performed here using two smoother algorithms, the forward backward smoother and the two-filter smoother. The approach is demonstrated on experimental data collected during both the SWellEx-96 and SW06 experiments where the uncertainty in the estimates is reduced.