## 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 $\mu$Pa.

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 $\mu$M) 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 $> 0.9$s 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.

4a THU. AM

**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

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"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

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 (OffNoise-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.