# Session 1aAA

## Architectural Acoustics: General Topics in Architectural Acoustics

Steven D. Pettyjohn, Chair
*The Acoustics & Vibration Group, Inc., 5700 Broadway, Sacramento, CA 95820*

### *Contributed Papers*

9:00

**1aAA1. Toward reliable metrics for Sacred Harp singing spaces.** Benjamin J. Copenhaver, Scott J. Schoen, and Michael R. Haberman (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, bcopenhaver@utexas.edu)

Sacred Harp singing, a common type of shape-note singing, is a centuries-old tradition of American community choral music. It is traditionally a participatory form of music with no distinction between performers and audience, a characteristic that makes for acoustical requirements that differ considerably from those of a concert hall or even a typical worship space. In the spirit of the text *Concert Halls and Opera Houses* by L. Beranek, we seek to correlate acoustical measurements of spaces used for Sacred Harp singing with subjective evaluations of those spaces made by the singers themselves. To achieve this, measurements of reverberation time and support factor of each space are coupled with participant surveys in 10 different Sacred Harp singing locations. Those measurements are then examined for their applicability as metrics for evaluation of Sacred Harp performance spaces. In addition, various measurement techniques for this type of space are explored and reported.

9:15

**1aAA2. The effect of two different rooms on acoustical and perceptual measures of mixed choir sound.** Kathryn S. Hom (1623 Alcatraz Ave. Apt. D, Berkeley, CA 94703, khom@ku.edu)

The purpose of this study was to explore the effect of two different rooms (choir rehearsal room, performance hall) on acoustical (LTAS, one-third octave bands) and perceptual (singer [$N = 11$] survey, listener [N = 33] survey, and Pitch Analyzer 2.1) measures of soprano, alto, tenor, and bass (SATB) choir sound. Primary findings of this investigation indicated: (a) significant differences in spectral energy comparisons of choir sound between rooms, (b) choristers' perceptions of hearing and monitoring their own voices differed significantly depending on room, (c) most choristers (82%) perceived that the choir performed best within the Performance Hall, (d) perceived pitch of selected sung vowels within recordings differed significantly based on room conditions, (e) 97% of listeners perceived a difference in choir sound between room recordings, and (f) most listeners (91%) indicated preference for the Rehearsal Room recording.

9:30

**1aAA3. Measuring and optimizing clarity in large and small spaces.** David H. Griesinger (David Griesinger Acoustics, 221 Mt. Auburn St. #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Psychologists know that sounds perceived as close to a listener hold attention and are easier to parse and to remember than sounds perceived as further away. But current measures for clarity are blind to the vital importance of sonic closeness in the transfer of information in enclosed spaces. Recent work in the neurology of the inner ear is illuminating the mechanisms by which the ear separates simultaneous sounds from separate sources, as well as the "closeness" of each source. Knowledge of these mechanisms allows us to predict how this ability is lost in the presence of reflections and noise, and to predict a number of ways that real clarity can be measured and optimized in classrooms, lecture halls, and performance venues of all types. This paper will describe and demonstrate how reflections degrade the closeness or clarity of sounds, and how this degradation can be prevented or ameliorated. Examples of old and new spaces with either excellent or poor clarity will be presented, along with a few examples of recent improvements to existing halls.

09:45

**1aAA4. The maximum intelligible range of the unamplified human voice.** Braxton B. Boren and Agnieszka Roginska (Music, New York Univ., 35 W. 4th St., New York, NY, bbb259@nyu.edu)

The Anglican preacher George Whitefield preached to some of the largest reported crowds in recent history during the Methodist revivals in 18th century London. Benjamin Franklin later performed an auditory experiment in Philadelphia from which he estimated Whitefield could be heard by 30,000 listeners at once. Using the data from Franklin's experiment and acoustic model of colonial Philadelphia, Whitefield's on-axis averaged sound pressure level at one meter has been calculated to be about 90 dBA, consistent with the loudest values measured from trained vocalists today. Using period maps and topological data, acoustic models have been constructed of the sites of Whitefield's largest crowds in London, using a human voice source with the projected SPL for Whitefield's preaching voice. Based on the total audience area whose speech transmission index value is greater than that at Franklin's position in the Philadelphia experiment, the total intelligible audience area can be calculated. Using Franklin's own crowd density calculations, this method allows estimates of the maximum amount of listeners that could hear Whitefield's voice under different environmental conditions and provides a better maximum estimate for the free-field intelligible range of the unamplified human voice.

10:00

**1aAA5. Headphone- and loudspeaker-based concert hall auralizations and their effects on listeners' judgments.** Samuel Clapp (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, clapps@rpi.edu), Anne E. Guthrie (Arup Acoust., New York, NY), Jonas Braasch, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Room impulse responses were measured in a wide variety of concert and recital halls throughout New York State using a spherical microphone array and dummy head as receivers. These measurements were used to create auralizations for second-order ambisonic playback via a loudspeaker array and headphone playback, respectively. The playback methods were first evaluated objectively to determine how accurately they could reproduce the measured soundfields with respect to spatial cues. Subjects were then recruited for listening tests conducted with both reproduction methods and asked to evaluate the different spaces based on specific parameters and overall subjective preference. The results were examined in order to determine

the degree to which judgments of the different parameters were affected by the playback method.

**10:15–10:30 Break**

**10:30**

**1aAA6. The effect of using matched, but not individualized, head related transfer functions on the subjective evaluation of auralizations.**
Matthew Neal and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., Appl. Sci. Bldg, University Park, PA 16802, mtn5048@psu.edu)

Head related transfer functions (HRTFs) are a key component when creating auralizations used in subjective concert hall studies. An average HRTF, rather than an individualized HRTF, is often used when creating auralizations, which can lead to front-back ambiguity and reduced out-of-head sound localization. The goal of this study was to determine how the choice of HRTF can impact subjective impression of various concert hall acoustic qualities from auralizations. Using 10 HRTFs from the CIPIC database, the best and worst case HRTFs were determined for each test subject, based on out-of-head localization. For the main test, auralizations were created using these two HRTFs, along with an average HRTF. The subjects' task was to evaluate the auralizations based upon concert hall acoustics characteristics, such as reverberance and listener envelopment. Before participating in the main part of the study, the subjects were required to complete a series of training sessions. Test subjects were limited to trained musicians and those who met a 15 dB hearing level requirement. The resulting data were analyzed to determine the significance of the differences between the three cases. The results from the listening tests and the statistical analysis will be presented. [Work supported by NSF Grant 1302741.]

**10:45**

**1aAA7. How room acoustics impact speech comprehension by listeners with varying English proficiency levels.**
Zhao Peng, Adam M. Steinbach, Kristin E. Hanna, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182, zpeng@huskers.unl.edu)

The authors previously reported the preliminary results from an investigation on effects of reverberation and noise on speech comprehension by native and non-native English-speaking listeners at ICA, Montreal. The results showed significant main effects of reverberation time (from 0.4 to 1.2 s) and background noise level (three settings of RC-30, 40, and 50). Non-native listeners performed significantly worse than natives on speech comprehension in general. Furthermore, the negative effect of reverberation was more detrimental for non-native than for native English-speaking listeners. However, the preliminary analyses have not yet accounted for effects of English proficiency on speech comprehension in addition to the room acoustic environments tested. All test participants were screened for three measures of English proficiency (listening span, oral comprehension, and verbal abilities). Non-native listeners as a group scored lower on all three proficiency measures than native English-speaking listeners. In this paper, English proficiency is investigated as confounding factor affecting speech comprehension performance alongside noise and reverberation. The results help to further the understanding of how room acoustics impacts speech comprehension by listeners, native and non-native English-speaking, with varying English proficiency levels. [Work supported by a UNL Durham School Seed Grant and the Paul S. Veneklasen Research Foundation.]

**11:00**

**1aAA8. Human responses to impulsive noises with accompanying rattle.**
Andrew Hathaway, Kristin Hanna, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska – Lincoln, 2554 N. 48th St., Omaha, NE 68104, ahathaway@unomaha.edu)

This research aims to quantify human reactions to short impulsive noise bursts with a rattle element, to complement research at NASA Langley Research Center on evaluating human response inside buildings to low-level sonic booms. While previous research has demonstrated effects of noise bursts of varying amplitudes on assorted cognitive tasks, more information is needed to indicate at what level and to what degree such noise bursts may impact human performance and perception. It is also thought that the addition of rattle, produced from the noise burst exciting assorted elements inside the building, could prove more detrimental to human responses than noise bursts alone. Seventeen participants each completed twelve 30-min sessions during which they were subjected to controlled yet randomly occurring 250 ms broadband noise bursts, either with or without a rattle component, while completing an arithmetic task utilizing working memory. Four different levels of noise bursts, from 55 to 70 dBA, were tested; accompanying rattle components were set to be 4 dBA louder than the associated burst, generated by a separate audio source. Results will be presented and compared to those from a similar test using just broadband noise bursts alone. [Work supported by a NASA Nebraska Space Grant.]

**11:15**

**1aAA9. The do's and don'ts of electronic sound masking.**
Roman Wowk (Papadimos Group, 300 Montgomery St., Ste. 908, San Francisco, CA 94104, roman@papadimosgroup.com)

Electronic sound masking, when properly implemented, can be an effective tool for introducing background noise in occupied spaces, which is a key factor of speech privacy. Mechanical ventilation systems generally cannot be relied upon to provide the consistent and appropriate levels of background noise that sound masking systems are capable of, and this is even more the case with variable-air-volume systems and several emerging (or re-emerging) technologies such as natural ventilation, underfloor air distribution, and chilled beams. However, despite general industry consensus of what constitutes appropriate levels of background noise, striking discrepancies in acoustic performance have been observed for several recent sound masking installations, thus defeating the original objectives for such a system. Fortunately, such issues can be avoided if design efforts account for limitations of both the sound masking system and acoustics of the space, clear performance requirements are established, and the system is properly adjusted and balanced prior to occupancy. Through case studies, this paper seeks through to illustrate the need for a guided approach toward achieving successful outcomes while also highlighting the broader range of issues involved with speech privacy.

**11:30**

**1aAA10. The acoustics of the catacombs of "San Callisto" in Rome.**
Amelia Trematerra and Gino Iannace (Dept. of Architecture and Industrial Des., borgo san lorenzo, Aversa 83016, Italy, amelia.trematerra@unina2.it)

The present study shows the acoustic properties of the catacombs of "St. Callisto" in Rome. Same acoustic measurements were made to understand which type of religious functions could have been celebrated in them, in particular, if they were spoken or sung. The catacombs were places of burial, but they were also places of reunions and visits to the graves of the martyrs. The catacombs are places for burial, diffused in the different religions of the different zones of Europe and Mediterranean Asia, not only Christian, even if they reached the maximum diffusion with the Christianity, the presence of the catacombs in these places it is due to the fact that the subsoil is easily excavatable. The catacombs were generally built through the realization of burrows, with corridors that connected all the "cubicles," rooms with a regular plant in which the burial happened. The acoustic measurements were carried out with an explosion of a balloon toy, with a microphone connected to a computer were measured the impulse response. The microphones were in different points of the corridors and in the "cubicles." The T30 measured values is short, for the presence the catacombs tuff walls.

## Session 1aAB

## Animal Bioacoustics: Passive Acoustic Monitoring of Marine Mammals

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

**Chair's Introduction—9:55**

### *Contributed Papers*

**10:00**

**1aAB1. Localization of humpback whale signals with two directional frequency analysis and recording sonobouys.** Alexis B. Rudd, Whitlow Au (Alexis Rudd, Hawaii Inst. of Marine Biol., 47-420 Hui Iwa St. #B-304, Kaneohe, HI 96158, rudd@hawaii.edu), Eva Marie Nosal (Alexis Rudd, Hawaii Inst. of Marine Biol., Honolulu, HI), and Seibert Murphey (Guide Star Engineering, Kapolei, HI)

For many years, the navy has used directional frequency analysis and recording (DIFAR) sonobouys to record and track ships. DIFAR sonobouys compute acoustic particle velocity for two bimodal perpendicular hydrophone elements, which, with a magnetic compass, gives a directional bearing to the sound recorded on a third omnidirectional hydrophone. Use of DIFAR represents an easily deployable alternative method to standard time-difference of arrival localization with towed or bottom-mounted hydrophone arrays. In this study, we used two tethered DIFAR sonobouys to record and track playbacks of humpback whale song. GPS positions of playbacks were recorded. Digital signals were recorded at the sonobouy before multiplexing. Song units were recorded on both sonobouys, and then matched between sonobouys using time from the sonobouy GPS. To localize the signals, azimuths from the DIFAR sonobouys were combined with the time difference of arrival parabola to create a three-dimensional likelihood surface that gives the probabilities that a signal originated at a specific point in space. The locations with maximum likelihood were used to estimate source level of whale calls, and singing whales were tracked over time. Comparisons will be made between DIFAR methods and GPS locations from the playback boat, to ground-truth this localization methodology.

**10:15**

**1aAB2. Highlighting pros and cons of abundance estimation using passive acoustic data: monitoring fin whales (*Balaenoptera physalus*) off the southern Portuguese coast using seismometers.** Danielle Harris, Tiago Marques (Ctr. for Research into Ecological and Environ. Modelling, Univ. of St Andrews, The Observatory, Buchanan Gardens, St Andrews KY16 9LZ, United Kingdom, dh17@st-andrews.ac.uk), Luis Matias (Instituto Dom Luiz, Universidade de Lisboa, Lisboa, Portugal), David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR), Elizabeth T. Küsel (Northwest Electromagnetics and Acoust. Res. Lab., Portland State Univ., Portland, OR), and Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St Andrews, St Andrews, United Kingdom)

Monitoring marine mammals using passive acoustic sensors is increasingly popular. Generating abundance estimates from acoustic data would be extremely useful for marine environment stakeholders. To achieve accurate abundance estimates, there are three broad areas to consider: (1) survey design, (2) data collection and processing, and (3) data analysis. Here, we use an analysis of fin whale (*Balaenoptera physalus*) calls recorded on ocean bottom seismometers (OBSs) to discuss the main advantages, disadvantages and considerations of abundance estimation using acoustic data. The OBS array was deployed for one year (2007–2008) and demonstrates how an opportunistic dataset can meet survey design requirements. Ranges to detected calls (detected with a matched filter) were estimated using the seismological three-component method. Point transect sampling, an abundance estimation method, was then used to estimate average call density. Animal density or abundance could not be estimated because the appropriate average calling rate was unknown. Finally, spatiotemporal patterns of call density were modeled. This dataset has also allowed new methods development—a method that estimates abundance as a function of total energy in a species' frequency band has been developed. In summary, abundance estimation using acoustic data is possible but challenging, and improved knowledge of vocal behavior is essential.

**10:30**

**1aAB3. The acoustic presence of sperm whales (*Physeter macrocephalus*) at Ocean Station PAPA in the Gulf of Alaska 2007–2012.** Nikoletta Diogou (Dept. of Marine Sci., Univ. of the Aegean, University Hill, Mytilene 81100, Greece, nikitsoto01@hotmail.com), Holger Klinck (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), and Jeffrey A. Nystuen (Appl. Phys. Labo., Univ. of Washington, Seattle, WA)

Sperm whale (*Physeter macrocephalus*) populations were severely depleted by commercial whaling worldwide in the 18th through the 20th century. Consequently, in 1970, this species was listed in the United States as an endangered species. To date, accurate information on the abundance and distribution of sperm whales in offshore areas of the North Pacific are scant. Sperm whales regularly produce high intensity sounds for navigation, prey detection, and communication. Thus, this species can be very effectively monitored using passive acoustic techniques especially in remote and inaccessible locations such as the Gulf of Alaska (GOA). In this study, a Passive Aquatic Listener (PAL) was deployed at Ocean Station PAPA (50°N, 145°W) in the GOA between 2007 and 2012 to monitor the seasonal occurrence of sperm whales in the area. Preliminary results indicate that within the 5-year deployment period sperm whales were acoustically present year round and that the number of acoustic sperm whale detections showed a seasonal trend with slightly higher numbers during the summer months. We are currently investigating the linkage between the occurrence of sperm whales and environmental conditions (e.g., Pacific Decadal Oscillation index) in the study area. [Funding from the Office of Naval Research.]

**10:45**

**1aAB4. Temporal and spatial trends in acoustic detections of delphinid species around Niihau, Hawaii.** Julie N. Oswald (Bio-Waves, Inc., 364 2nd St., Ste. #3, Encinitas, CA 92024, julie.oswald@bio-waves.net), Whitlow W. L. Au, Marc O. Lammers, Michael F. Richlen (Hawaii Inst. of Marine Biol., Univ. of Hawaii, Kaneohe, HI), and Thomas F. Norris (Bio-Waves, Inc., Encinitas, CA)

Passive acoustic monitoring using seafloor-mounted recorders allows cetacean occurrence to be examined over time and space. Four ecological

acoustic recorders (EARs) were moored around the Hawaiian island of Niihau in summer/fall (July—November) 2011, and winter/spring (January—May) 2012. Delphinid whistle "detections" (a proxy for schools) were identified and characterized. Whistles were identified to species using a random forest classifier trained with whistles recorded from seven species (*Globicephala macrorhynchus, Pseudorca crassidens, Stenella attenuata, S. coeruleoalba, S. longirostris, Steno bredanensis,* and *Tursiops truncatus*) in the tropical Pacific Ocean. The highest number of detections per day occurred during summer/fall at all sites. All species except for *G. macrorhynchus* were detected at every site during both deployments. No single species dominated the detections at any site, with the exception of *Stenella longirostris* at the Pueo Point site during summer/fall (53% of detections). *Pseudorca crassidens*, a species of particular management/conservation interest due to small population sizes, were detected most frequently (18% of detections) at the Niihau NW site during summer/fall and least frequently (7% of detections) at the Pueo Point site during summer/fall. Understanding trends in species composition provides insight into how species use different habitats and aids in management efforts.

**11:00**

**1aAB5. Long term passive acoustics monitoring reveals differences in deep diving Odontocetes foraging strategies.** Giacomo Giorli (Dept. of Oceanogr., Univ. of Hawaii at Manoa, 1000 Pope Rd., Honolulu, HI 96822, giacomog@hawaii.edu) and Whitlow W. Au (Hawaii Inst. of Marine Biol., Univ. of Hawaii at Manoa, Kaneohe, HI)

Beaked whales, sperm whales, pilot whales, and Risso's dolphins perform deep dives to feed on prey in the deep sea. They use echolocation to detect prey, and echolocation can be monitor as a proxy of their foraging activities. Ecological acoustic recorders were deployed to monitor in time the echolocation signals of pilot whales, Risso's dolphins, beaked whales, and sperm whales near Josephine Seamount (Portugal), in the Ligurian Sea (Italy), and along the Kona coast of the island of Hawaii. Data analysis was performed using two automatic detector/classification systems: Marine Mammal Monitoring on Navy Ranges (M3R), developed at the Naval Undersea Warfare Center Division in Newport, USA, and a custom MATLAB program. An operator-supervised custom MATLAB program was used to validate the classification performance, which was higher than 85% for each category. Results show that pilot whales and Risso's dolphins concentrate their feeding effort mainly during the night. The foraging activity of beaked and sperm whales is variable, and different patterns are observed at different locations. Prey behavior might play a central role in driving the foraging activity of deep divers, but to date, the reasons why such activity varies between species that might feed on similar food resources remains unknown.

**11:15**

**1aAB6. Transmission characteristics of Arctic pinniped vocalizations through ice.** Samuel L. Denes (Graduate Program in Acoust., Penn. State Univ., 116 Appl. Sci. Bldg., University Park, PA 16802, sld980@psu.edu), Carl Hager (Dept. of Oceanogr., United States Naval Acad., Annapolis, MD), and Jennifer L. Miksis-Olds (Appl. Res. Lab., The Penn State Univ., University Park, PA)

Arctic pinniped vocalizations occur in an environment where acoustic coupling between water, air, and ice must be considered in propagation effects. A low amplitude propagation experiment was conducted to measure the transmission characteristics of water borne acoustic signals through ice and into air. The signals used were inspired by arctic pinniped vocalizations. Signals included impulses, frequency modulated downsweeps, and continuous frequency tones. An array of 12 elements including 6 hydrophones, 3 geophones, and 3 microphones were deployed off the coast of Barrow, Alaska, in March of 2012. Three stations consisting of two hydrophones, one geophone and one microphone were arranged linearly in one ice pan with a source speaker placed endfire and broadside. Ice thickness varied from 1.5 to 2.5 m between the receiver locations. Measured transmission characteristics of each of the signals are analyzed as a function of distance from the source and thickness of ice. Received levels are compared with estimates from a wavenumber integration propagation model. Communication ramifications from changing Arctic ice conditions will be discussed.

**11:30**

**1aAB7. A modified wavefront curvature method for the passive ranging of echolocating dolphins in the wild.** Eric L. Ferguson (Inst. of Marine Sci., Univ. of Sydney, Madsen Bldg., Sydney, New South Wales 2006, Australia, efer2780@uni.sydney.au)

Passive ranging by wavefront curvature is a practical nonintrusive method for studying the behavior of echolocating odontocetes (toothed whales including dolphins) in their natural habitats. This method requires two differential time-of-arrival measurements of the signal wavefront using three hydrophones, which are widely spaced along a straight line. However, if the middle hydrophone is displaced (even by a small amount) from an imaginary line connecting the other two (outer) sensors, then the source range estimate can be significantly biased. A modification to the method is shown to solve this problem so that the range bias is zero. The source locations of echolocating dolphins are estimated using the modified wavefront curvature method and they are found to match those estimated by other (line of position and circle intersection) methods. The application of the modified method to a series of dolphin click sequences demonstrates that the method is a powerful tool for studying the behavior of free-ranging echolocating dolphins in the wild. For example, individual dolphins are located with unprecedented accuracy even when multiple echolocating dolphins are present.

**11:45**

**1aAB8. Comparison of remote ranging techniques for bowhead whale calls in a dispersive underwater sound channel.** Shima Abadi (Mech. Eng., Univ. of Michigan, 2010 W.E.Lay Automotive Lab., 1231 Beal Ave., Ann Arbor, MI 48109, shimah@umich.edu), Aaron M. Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Locating and monitoring marine mammals near anthropogenic sound sources are important tasks with both environmental and operational consequences. This presentation describes how mode filtering (MF) or a blind-deconvolution technique (synthetic time reversal, STR) can be used to determine the range of bowhead whale calls from a single linear vertical array in a dispersive underwater sound channel. The results are based on environmental parameters at the array location, and simulations and ocean recordings of natural whale calls with a nominal bandwidth from 50 to 500 Hz. The passive listening experiments were conducted in the coastal waters near Kaktovik, Alaska, with a 12-element vertical array nominally spanning the middle 60% of the water column. It was deployed in 55-m-deep water alongside distributed arrays of Directional Autonomous Seafloor Acoustics Recorders (DASARs) arranged in triangular grids used to horizontally localize whale calls. A total of 18 naturally occurring whale calls were considered. The estimated call-to-array ranges determined from mode filtering and STR are compared with reference triangulation results from the DASARs. The vertical-array ranging results are generally within $\pm 10\%$ of the DASARs results with the STR results being slightly more accurate than those from mode filtering. [Sponsored by ONR and Shell.]

# Session 1aAO

## Acoustical Oceanography: Acoustics for Cabled Ocean Observatories

Thomas Dakin, Cochair
*Ocean Networks Canada, TEF-128A 2300 McKenzie Ave., Univ. of Victoria, Victoria, BC V8W2Y2, Canada*

Bruce M. Howe, Cochair
*Ocean and Resources Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 402, Honolulu, HI*

**Chair's Introduction—8:25**

### *Invited Papers*

**8:30**

**1aAO1. Protocols for acoustic devices on cabled ocean observatories.** Richard K. Dewey, Steve Mihaly, and Tom Dakin (Ocean Networks Canada, Univ. of Victoria, 2300 McKenzie Ave., Victoria, BC V8N 5M7, Canada, rdewey@uvic.ca)

Cabled ocean observatories offer an unprecedented opportunity for marine acoustic devices. With continuous power and high bandwidth, long-term broadband measurements from both passive and active systems are possible over a wide range of oceanographic environments. In particular, the cabled ocean observatories VENUS and NEPTUNE operated by Ocean Networks Canada offer a unique set of possibilities for acoustic research, ranging from littoral to mid-ocean ridge installations. Since 2006 fiber optic telecommunication cables have provided high power and up to GB data rates from nine distributed observatory Nodes across marine provinces, which include fjords, tidal straits, river deltas, continental shelves, slopes and rises, abyssal plains, and spreading margins. Experience from passive hydrophones as well as both mono- and bi-static active systems including echo-sounders, sonars, and many acoustic Doppler current devices will be discussed. This paper will review our experiences, provide examples, and propose protocols, or at least our suggested best practices, for many active and passive acoustic devices, including calibration, configuration, interference reduction techniques, bandwidth considerations, data handling, and data analysis and delivery methods.

**8:50**

**1aAO2. Hydrophone data management at Ocean Networks Canada.** Benoît Pirenne (Digital Infrastructure, Ocean Networks Canada, 2300 McKenzie Ave., Victoria, BC V8P 2N7, Canada, bpirenne@uvic.ca), John Dorocicz (Marine Operations, Ocean Networks Canada, Victoria, BC, Canada), and Tom Dakin (Centre for Innovation, Ocean Networks Canada, Victoria, BC, Canada)

In this contribution, we describe the seabed cabled observatory networks managed by Ocean Networks Canada (ONC). The focus is on the hydrophone hosting capabilities of ONC's advanced science infrastructure and on the assets already in place, together with their scientific objectives. We describe the data acquisition and archival principles and methods, as well as the derived products available to users. We also elaborate on the concerns of the United States and Canadian Navies with quasi real-time, public data and on the mitigation measures in place to balance security needs with the scientific and public safety goals of the instrumentation. Finally, we describe the potential for instrument development and testing that ONC offers on its networks.

**9:10**

**1aAO3. Remote performance assessment of cabled observatory hydrophone systems.** Nathan D. Merchant (Department of Physics, Univ. of Bath, Bath, United Kingdom), D. Tom Dakin, John Dorocicz (Ocean Networks Canada, Univ. of Victoria, Ocean Networks Canada, Victoria, BC, Canada, tdakin@uvic.ca), and Philippe Blondel (Dept. of Phys., Univ. of Bath, Bath, United Kingdom)

The increasing expansion of cabled undersea observatories worldwide presents a unique opportunity to develop and deploy hydrophone systems for long-term monitoring of underwater noise. Among the first such observatories, the NEPTUNE and VENUS networks operated by Ocean Networks Canada (ONC) have been pioneering in their implementation of acoustic monitoring systems and are projected to continue operation with an anticipated lifetime of 25 years. One challenge that arises from decades-long deployments is the need to assess the consistency of data quality from operational instruments. Since the replacement or *in situ* calibration of instruments can be costly and problematic, it is expedient if such performance assessment can be conducted remotely. Here, we present methods of detecting deterioration in the performance of cabled hydrophone systems and assessing the suitability of the dynamic range to the prevailing noise conditions. Approaches are proposed based on tracking variability in the primary mode, detection and classification of persistent tonal components, and automatic detection of dynamic range exceedance. Scenarios based on data from ONC observatories are presented.

9:30

**1aAO4. Measurement of hydrothermal heat flux using a sonar deployed on the Canadian Neptune cabled observatory.** Guangyu Xu, Peter A. Rona, Karen G. Bemis (Inst. of Marine and Coastal Sci., Rutgers Univ., New Brunswick, NJ), and Darrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, drj@apl.washington.edu)

The Cabled Observatory Vent Imaging Sonar (COVIS) was deployed at the Main Endeavour Field node of the Canadian NEPTUNE cabled observatory in September 2010 and has acquired long time series on plume and diffuse hydrothermal flows. This talk will focus on recent efforts by the Rutgers-APL collaboration to invert sonar data to determine heat flux from the Grotto plume complex. Inversion employs plume theory to relate velocity as determined by Doppler shift to buoyancy flux, hence heat flux. The primary uncertainties have to do with plume bending due to ambient current and short sampling times relative to dynamic changes in plume shape. These uncertainties have been quantified by means of special high-statistics experiments using COVIS. Time series for heat flux will be compared with ground truth obtained by thermometry using an ROV. [Work supported by NSF Grants OCE-0824612 and OCE-0825088.]

## *Contributed Papers*

9:50

**1aAO5. Acoustics at the ALOHA Cabled Observatory: On-going results and new instruments.** Bruce Howe and Ethan H. Roth (Ocean and Resources Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, bhowe@hawaii.edu)

Since June 2011, the ALOHA Cabled Observatory (ACO) has been collecting ocean acoustic data, continuing an earlier data set covering February 2007—October 2008. The ACO is at Station ALOHA 100 km north of Oahu, the field site of the Hawaii Ocean Time-series (HOT) program that has collected biological, physical, and chemical oceanographic data since 1988. At 4728 m water depth, it is the world's deepest operating cabled observatory. On-going acoustics results will be presented along with results from two new instruments to be deployed: a WHOI micro-modem, and a camera/hydrophone combination. Plans for future acoustics research will be discussed. [Work supported by the National Science Foundation.]

**10:05–10:20 Break**

10:20

**1aAO6. Seasonal and diurnal pattern of baleen whales detected on the Station Aloha Cabled Observatory off Oahu, Hawaii.** Whitlow W. Au, Michael Richlen, Hui Ou (Hawaii Inst. of Marine Biol., Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, wau@hawaii.edu), and Bruce Howe (Ocean and Resources Eng., Univ. of Hawaii, Honolulu, HI)

The ALOHA Cabled Observatory, ACO, located 100 km north of the island of Oahu, Hawaii, includes an acoustic recording package. The hydrophone is at a depth of 4.7 km and is connected to land via an underwater fiber optic cable. Recordings are continuous and made year-round. One year of data, 17 February 2007 to 18 February 2008, were analyzed to detect calls from blue (*Balaenoptera musculus*), fin (*Balaenoptera physalus*), sei (*Balaenoptera borealis*), and minke whales (*Balaenoptera acutorostrata*). As expected, detections of all four species were highly seasonal with most occurring during the winter months. While the exact migration routes are not well understood for these species, a pattern similar to the breeding/calving migrations of humpback whales would be anticipated. Fin whales were detected most often followed by minke, blue, and sei whales. No diurnal patterns were observed for the fin, sei, and minke whales; however, a diurnal pattern was evident for blue whales (59% of calls were made in the evening and nighttime hours). All of the species are very rarely seen in Hawaiian waters, and blue whales have never been detected during any visual surveys. Acoustics detections are essential for understanding the occurrence of these species in Hawaii.

10:35

**1aAO7. Contribution of iceberg sounds to the ambient noise budget in the South Pacific Ocean.** Haru Matsumoto (CIMRS, Oregon State Univ., 2115 SE OSU Dr., Newport, OR 97365, haru.matsumoto@oregonstate.edu), Del Bohnenstiehl (MEAS, North Carolina State Univ., Raleigh, NC), Robert Dziak, Joe Haxel (CIMRS, Oregon State Univ., Newport, OR), Minkyu Park, Won-Sang Lee (Korea Polar Res. Inst., Yeonsu-gu, Incheon, South Korea), Tai-Kwan Lau, and Matt Fowler (CIMRS, Oregon State Univ., Newport, OR)

On May 2002, C19, a 5500 $km^2$ iceberg calved from the Ross Ice Shelf, eventually drifting eastward into the open Pacific Ocean by 2008. As it sailed into warmer waters, thermal and wind stresses caused the iceberg to crack and break apart. These resulting "icequakes" projected wideband acoustic energy into the water column, influencing the regional ambient noise environment. Icequake noise was persistent and strong enough to be observed by NOAA's eastern equatorial Pacific moored hydrophone (EEP-NW at 8N, 110W) as well as the hydroacoustic station of International Monitoring System (IMS) on Juan Fernandez Island (H03N at 33.44S, 78.91W). Elevated noise levels (maximum of ~+3 dB at NOAA's EEP and ~+7 dB at IMS H03N hydrophones) were observed by both stations from early 2008 when C19a first appeared in the Pacific until it drifted into the Atlantic Ocean in early 2009. C19a's icequake and calving activity was also most frequent during this same period. Seasonal changes and long-term trends in ambient noise levels at NOAA's EEP-NW acoustic mooring (1996–2009) and IMS Juan Fernandez (2003–2010 years) and the unique acoustic role icebergs play in the Southern Ocean will be presented.

**10:50–11:50 Panel Discussion**

# Session 1aBA

## Biomedical Acoustics: Bubble Detection in Diagnostic and Therapeutic Applications I

Eleanor P. Stride, Chair
*Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom*

**Chair's Introduction—7:55**

### *Invited Papers*

**8:00**

**1aBA1. Single microbubbles vibrations measured with an "acoustical camera".** Guillaume Renaud, Johan Bosch, Antonius van der Steen, and Nico de Jong (Biomed. Eng., Erasmus MC, Dr. Molewaterplein 50, Rotterdam 3015 GE, Netherlands, n.dejong@erasmusmc.nl)

Measuring and understanding the nonlinear vibrations of contrast agent microbubbles in response to an ultrasound wave is essential to optimize the technique employed by an ultrasound scanner to distinguish microbubbles from tissue. In this context, an acoustical method was developed to retrieve the radial response of single microbubbles to a pressure wave by means of a low-amplitude probing wave. If the frequency of the latter is much higher than the spherical resonance frequency of the microbubble (typically between 1 and 10 MHz), the relative amplitude modulation (induced by a pressure wave) in the signal scattered in response to the probing wave is quasi-equal to the radial strain (i.e., relative variation in radius) induced by the pressure wave. A reference response to the probing wave acquired before and after the transmission of the pressure wave allows us to reveal asymmetry in microbubble oscillations. Although efficient nonlinear wave interaction is well known in bubbly liquids, we demonstrated that such an "acoustical camera" can extract quantitative information (the radius as function of time) on single bubble vibrations by analyzing the nonlinear coupling between two ultrasound waves.

**8:20**

**1aBA2. The mechanical index and bubbles in tissue, an evidentiary review.** Charles C. Church (National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, cchurch@olemiss.edu)

The mechanical index (MI) quantifies the likelihood that diagnostic ultrasound will produce an adverse biological effect by a non-thermal mechanism. The current formulation of the MI is based on inertial cavitation thresholds in water and blood as calculated for a pulse duration of one period. However, tissue is not a liquid but a viscoelastic solid, and imaging pulses may be much longer. The importance of these differences has been quantified using the Gilmore equation for water and blood and a form of the Keller-Miksis equation modified for soft tissue. It is shown that the threshold for cavitation is higher in soft tissues, and much higher in muscle, than in blood. Experimentally determined thresholds in tissue lie above these theoretical results by up to an order of magnitude. This suggests that use of the MI as part of the FDA 501(k) approval process may unnecessarily restrict the outputs of diagnostic ultrasound machines. Additionally, a simple analysis of another potential mechanism for biological effects, radiation force, indicates that the very form of the MI does not include all non-thermal mechanisms but, perhaps, only one. These results cast doubt on the value of the MI as a reliable means of assessing patient safety.

**8:40**

**1aBA3. Transcranial bubble activity mapping for therapy and imaging.** Meaghan A. O'Reilly (Physical Sci. Platform, Sunnybrook Res. Inst., 2075 Bayview Ave., Toronto, ON M4N 3M5, Canada, moreilly@sri.utoronto.ca), Ryan M. Jones, and Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Bubble-mediated ultrasound therapies in the brain, such as targeted disruption of the blood-brain barrier (BBB) or cavitation-enhanced stroke treatments, are being increasingly investigated due to their potential to revolutionize the treatment of brain disorders. Due to the fact that they are non-thermal in nature, these therapies must be monitored by acoustic means to ensure efficacy and safety. A sparse, 128-element hemispherical receiver array (612 kHz) was integrated within a 306 kHz therapy array. The receiver arrangement was optimized through numerical simulations. The array was characterized on the benchtop to map the activity of bubbles in a tube phantom through an *ex vivo* human skullcap. *In vivo* the array was used to map bubble activity in small animal models during microbubble-mediated BBB disruption. The array was investigated as well for diagnostic purposes, imaging transcranial structures filled with very dilute concentrations of microbubbles. A spiral tube phantom with tube diameter of 255 μm was imaged, using a non-invasive phase correction technique, through an *ex vivo* human skullcap by mapping the activity from single bubbles. Applying super-resolution techniques, an image of the spiral phantom was produced that was comparable to an image obtained in a small-specimen micro CT.

9:00

**1aBA4. Acoustical resonators: A versatile tool for bubble detection.** Helen Czerski (Inst. for Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton SO17 1BJ, United Kingdom, h.czerski@soton.ac.uk)

It can be tricky to quantify bubble size distributions accurately, especially for broad distributions of sub-millimeter bubbles. One solution to the problem is an acoustical resonator, a device which has been developed over the past two or three decades for use in the ocean. A single broadband measurement can provide a detailed bubble size distribution for bubbles from 5 microns to 500 microns in radius. Resonators are physically robust, provide data which needs little post-processing, and they have the potential for use in many other situations. In the ocean, we have used it to understand bubble coatings, follow bubble dissolution, and monitor void fraction in real time. I will describe the current state of resonator technology, show ocean data, and discuss its potential advantages and disadvantages for use in other environments.

## *Contributed Papers*

9:20

**1aBA5. An ultrasound system to identify and characterize kidney stones.** Bryan W. Cunitz, Barbrina L. Dunmire (Ctr.Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Mathew D. Sorensen, Ryan Hsi, Franklin Lee (Dept. of Urology, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105), Oleg A. Sapozhnikov (Acoust. Dept. and Appl. Phys. Lab., Moscow State Univ. and Univ. of Washington, Seattle, WA), Jonathan D. Harper (Dept. of Urology, Univ. of Washington, Seattle, WA), and Michael Bailey (Ctr.Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, bailey@apl.washington.edu)

Ultrasound imaging has tissue and blood imaging modes. This report describes development of a kidney stone imaging mode. Two plane pulses generate a B-mode image. Overlaid in color are regions of high decorrelation between the pulses. Our previous data [UMB, **39**, 1026–1038 (2013)] indicate the pulses excite bubbles on the stone surface, which causes the decorrelation. As such this mode automatically identifies stones in the image while scanning at a high frame rate. Further in a control box placed on the stone, highly focused beams are scanned across the stone and a harmonic B-mode image is produced to sharpen the lateral resolution. This mode is used to refine the size and shape of the stone. The first mode is used to aid visualization of stones. Our team is also using it to target and track stones that move with respiration during shock wave lithotripsy (SWL) and as an indicator of stone susceptibility to SWL since surface bubbles contribute to comminution. Improved stone sizing by the second mode aids treatment planning, and resolution of surface roughness is another indicator of stone fragility. [Work supported by NIH DK043881, NIH DK092197, and NSBRI through NASA NCC 9-58.]

9:35

**1aBA6. Twinkling artifact of Doppler imaging for cavitation detection during high-intensity focused ultrasound therapy: Sensitivity and resolution.** Tong Li, Tatiana D. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 7312 Linden Ave. N, Unit 7312, Seattle, WA 98103, tongli@u.washington.edu), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Moscow, Russian Federation), Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Seattle, WA), and Joo Ha Hwang (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

The mechanism of the twinkling artifact (TA) during Doppler imaging of kidney stones is well-known and is hypothesized to stem from the irregular scattering of Doppler ensemble pulses from fluctuating microbubbles trapped in crevices of the kidney stone. We have previously demonstrated that the TA can be used to detect and image microbubbles in soft tissue during pulsed HIFU treatment. In this work, the characteristics of the method—the sensitivity to small bubbles and the spatial resolution—were investigated experimentally and compared to other passive and active cavitation imaging methods such as pulse inversion. An approach was proposed for quantification of the cavitation images provided by the method, and the resulting metric was compared to the inertial cavitation dose. The experiments were performed using pulsed 1-MHz HIFU exposures of transparent gel phantoms, *ex vivo* tissue and *in vivo* mouse model of pancreatic cancer. [Work supported by RFBR and NIH (EB007643, 1K01EB015745, and R01CA154451).]

9:50

**1aBA7. Dual-frequency intravascular ultrasound transducer design for contrast agent imaging.** K. Heath Martin (The Joint Dept. of Biomed. Eng., The Univ. of North Carolina and North Carolina State Univ., 109 Mason Farm Rd., 310 Taylor Hall, Chapel Hill, NC 27599, khmartin@ncsu.edu), Jianguo Ma, Xiaoning Jiang (Mech. & Aerosp. Eng., North Carolina State Univ., Raleigh, NC), and Paul A. Dayton (The Joint Dept. of Biomed. Eng., The Univ. of North Carolina and North Carolina State Univ., Chapel Hill, NC)

Intravascular ultrasound (IVUS) is a unique diagnostic tool for assessing the composition and degree of stenosis of atheromata in cardiovascular disease. As of yet, contrast enhanced IVUS has not been clinically implemented despite its potential to provide functional information, such as degree of plaque vascularization, in conjunction with standard anatomical mapping. In this study, we demonstrate that nonlinear microbubble signals can be detected with minimal signal processing using an ultra-broadband IVUS transducer composed of a low frequency transmit and a high frequency receive combined element. Raw signals were acquired from a single element transducer ($1 \times 3$ mm, $f_c = 6.5/35$ MHz) after aligning to an acoustically transparent 200 μm inner diameter cellulose tube. Poly-disperse, (1–10 μm diameter) lipid-coated microbubbles were injected at high concentration ($4.8 \times 10^8$ MBs/mL) and peak negative pressures were varied from $-0.3$ to $-1.8$ MPa while transmission cycles were increased from 1 to 5. *In vitro* results indicate signal to noise ratios of 11 dB are attainable at 2$^{nd}$ harmonics and similar SNRs (10–8 dB) at higher harmonics. The results of this study indicate that contrast enhanced IVUS imaging is possible using a single transmit pulsing scheme enabled by using a dual- frequency broadband transducer design.

10:05–10:30 Break

10:30

**1aBA8. A cavitation detector for microbubble therapy based on the Stockwell transform.** Charles F. Caskey (Inst. of Imaging Sci., Vanderbilt Univ. 1161 21st Ave. South, MCN R0101, Nashville, TN 37203, cfcaskey@ucdavis.edu), Dustin Kruse, and Katherine W. Ferrara (Biomed. Eng., Univ. of California at Davis, Davis, CA)

Quantification of the wideband signal associated with bubble collapse is desirable to measure cavitation dosage during microbubble therapy. We have developed a cavitation detection algorithm based on time-frequency analysis with the Stockwell transform. This transform is ideal since it isolates cavitation transients in time and frequency at much higher resolution than other time-frequency methods. In this study, we acquired simultaneous high-speed images and acoustic signals from bubbles undergoing oscillation during acoustic pulses commonly used for permeability enhancement (Transmit: 1.5 MHz, 10-cycle, peak negative pressure range 150–500 kPa, Receive: 15 MHz cavitation detector). Above 200 kPa, linescan vs time images showed relative bubble expansion greater than 2 and echoes contained a wideband signal that coincided with bubble collapse. The Stockwell transform of detected echoes revealed wideband spectrums localized to the time of bubble collapse. After filtering below 4 MHz to reject linear echoes, the root mean square (RMS) of the spectral amplitude was obtained from the time-frequency decomposition, yielding peaks at cavitation transients. The magnitude of the Fourier transform of the RMS signal at 1.5 MHz provided a sensitive indicator for bubble

collapse and differed significantly across pressures where cavitation occurred ($p < 0.05$,$t$-test). The proposed measurement provides a sensitive metric for cavitation activity during microbubble therapy.

**10:45**

**1aBA9. Acoustic angiography ultrasound imaging can distinguish vascular differences in tumor cell lines.** Sarah E. Shelton (Biomed. Eng., UNC-NCSU Joint Program, 304 Taylor Hall, CB 7575, Chapel Hill, NC 27599, sarahshelton@unc.edu), James M. Dunleavey, Andrew C. Dudley (Cell and Molecular Physiol., UNC, Chapel Hill, NC), and Paul A. Dayton (Biomed. Eng., UNC-NCSU Joint Program, Chapel Hill, NC)

Acoustic Angiography is a novel ultrasound imaging technology for visualizing intravascular microbubble contrast agents. It uses a prototype dual-frequency transducer to transmit ultrasound pulses at a low frequency near microbubble resonance, and confocally receives higher frequency signal from the broadband microbubble response. By utilizing two widely separated frequencies (transmit at 4 MHz, receive at 30 MHz), it is possible to form a high-resolution image of the vascular structure without any tissue background. Both Acoustic Angiography and standard amplitude-modulation, nonlinear contrast perfusion imaging were used to compare the vascularity of xenograft tumors composed of two tumor cell sub-types with known differences in vascularity. In both image types, tumors were manually segmented in three dimensions and the percent of pixels containing contrast signal was computed. Acoustic Angiography imaging proved to be more sensitive to the vascular differences than standard nonlinear perfusion imaging. For a population of 16 animals, the sensitivity of Acoustic Angiography was 0.73, whereas the sensitivity of non-linear perfusion imaging was 0.13, given a significance level of 0.05. Additionally, two-sided $t$-tests showed a statistically significant difference between tumor types for Acoustic Angiography images ($p = 0.0025$), but not for standard nonlinear contrast imaging ($p = 0.39$).

**11:00**

**1aBA10. The late onset of nonlinear emissions from an ultrasound contrast agent.** Himanshu Shekhar (Elec. and Comput. Eng., Univ. of Rochester, 345 Hopeman Bldg., University of Rochester River Campus, Rochester, NY, himanshuwaits@gmail.com), Joshua J. Rychak (Targeson Inc., San Diego, CA), and Marvin M. Doyley (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

Microbubble contrast agents (MCA) are currently being investigated for several diagnostic and therapeutic applications. The nonlinear emissions of MCA are important for imaging and therapy; therefore, they need to be thoroughly characterized. In this study, we to investigated the temporal changes in the nonlinear emissions from a phospholipid-coated MCA. We characterized the nonlinear emissions over 60 min at 10 MHz, with 64 cycle excitation pulses, 30—290 kPa pressures, and 1 kHz pulse repetition frequency. The second harmonic, subharmonic, and ultraharmonic response were measured relative to the fundamental signal from the agent backscatter spectra. It was observed that emissions at the second harmonic did not change appreciably over the measurement period. However, the emissions at the subharmonic and ultraharmonic frequency increased by 18 and 8 dB over 20 min (relative to measurements at $t = 0$) and then decreased slightly. The highest relative increase was observed within the first 5 min —a timescale that can be relevant for clinical applications. These findings could help avoid errors in quantitative estimates obtained from nonlinear imaging, pressure estimation, and therapy monitoring techniques. Further, an elaborate understanding this phenomenon could help design efficacious MCA for clinical applications.

**11:15**

**1aBA11. Observed cumulative time delay between second harmonic and fundamental component of pressure wave fields propagating through ultrasound contrast agents.** Libertario Demi, Giovanna Russo (Eindhoven Univ. of Technol., Den Dolech 2, Eindhoven 5612 AZ, Netherlands, l.demi@tue.nl), Hessel Wijkstra (AMC Univ. Hospital, Amsterdam, Netherlands), and Massimo Mischi (Eindhoven Univ. of Technol., Eindhoven, Netherlands)

To our knowledge, despite several studies on the propagation velocity of pressure wave fields through ultrasound contrast agents (UCAs) have been conducted, the variation of propagation velocity between the fundamental and second harmonic component generated during the propagation of ultrasound through UCAs has not been studied yet. For this purpose, transmission and backscattering measurements of pressure wave fields propagating through gelatin phantoms containing cylindrical cavities of different diameter, filled with different concentrations [0 to 240 $\mu$L/L] of SonoVue® contrast agent, are conducted. Different frequencies [2, 2.5, and 3 MHz] and mechanical indices [0.05, 0.1, and 0.2] are investigated. Moreover, measurements obtained with Definity® contrast agent are also analyzed. Results show the occurrence of a cumulative delay between the time signals related to the second harmonic and fundamental component, suggesting a smaller propagation velocity for the second harmonic as compared to the fundamental component. Moreover, this delay increases with increasing microbubble concentration and diameter of the cavity, it depends on mechanical index and frequency, and, most importantly, it is not observed in the absence of UCAs. These results may be relevant for contrast-enhanced ultrasonography, providing new possibilities to increase contrast-to-tissue ratios and to quantify UCA concentrations.

**11:30**

**1aBA12. Effects of contrast agents and high intensity focused ultrasound on tumor vascular perfusion.** Linsey C. Phillips, K. Heath Martin, Ryan C. Gessner, and Paul A. Dayton (Joint Dept. of Biomed. Eng., UNC-Chapel Hill and NC State Univ., CB 7575, Univ. of North Carolina, Chapel Hill, NC 27599, linsey@email.unc.edu)

High intensity focused ultrasound (HIFU) is a treatment modality which causes localized tissue ablation and is further enhanced when microbubble or nanodroplet contrast agents are present. While it is known that HIFU ablation results in diminished blood perfusion, these effects have never been mapped with high-resolution 3-D imaging. We aimed to longitudinally and volumetrically evaluate the effects of microbubbles and nanodroplets on tumor vessel perfusion following HIFU treatment. Rats bearing flank fibrosarcoma tumors were injected with $1.5 \times 10^9$ microbubbles ($n = 3$), nanodroplets ($n = 3$), or control/no-injection ($n = 3$) during targeted HIFU (125 W/cm$^2$, 1 MHz, 2 MPa, 10% duty cycle, 1 Hz PRF, 3 min). This HIFU power is ~99% less than clinical norms. Tumor perfusion was assessed by contrast enhanced 3-D acoustic angiography before, immediately following, and 72 h after HIFU application. No significant changes in tumor perfusion were observed when HIFU was applied with nanodroplet or control injections. However, HIFU in the presence of microbubbles resulted in a $66.2 \pm 15.2\%$ ($p < 0.05$) reduction in perfused tumor volume. These results suggest that HIFU, when combined with microbubbles, can dramatically remodel tumor vasculature networks even at the low pressure used in this study. This could potentially enhance the safety of HIFU therapy.

**11:45**

**1aBA13. Modeling the loss of echogenicity from ultrasound contrast agents.** Kenneth B. Bader (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC 3935, Cincinnati, OH 45267-0586, Kenneth.Bader@uc.edu), Kirthi Radhakrishnan (Biomed. Eng. Program, Univ. of Cincinnati, Cincinnati, OH), Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Jason L. Raymond (Biomed. Eng. Program, Univ. of Cincinnati, Cincinnati, OH), Shao-Ling Huang, Tao Peng, David D. McPherson (Dept. of Internal Medicine, Univ. of Texas Health Sci. Ctr., Houston, TX), and Christy K. Holland (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Acoustically induced loss of echogenicity (LOE) from ultrasound contrast agents (UCAs) has been exploited in imaging techniques to improve delineation of pathology. Determination of the type of cavitation that accompanies LOE can be experimentally difficult to determine due to the complex microbubble activity elicited. A theoretical model has been derived to predict the LOE originating from rupture of the UCA shell, from stable cavitation, or from inertial cavitation. The predictions of the model for each cavitation phenomena will be compared to recent experimental LOE measurements of the lipid-based UCA Definity® and echogenic liposomes insonified by Doppler pulses from a clinical scanner. The backscatter coefficient was calculated for a population of UCAs exposed to 6-MHz pulsed ultrasound of duration 1.67 $\mu$s—8.33 $\mu$s. The change in the total backscatter coefficient was used to predict the LOE. The size distribution of UCAs was adjusted according to the specific type of cavitation triggered by the ultrasound exposure. Comparison of the theoretical predictions and experimental measurements suggest that shell rupture is the dominant mechanism for LOE for both Definity® and echogenic liposomes. These results will be discussed in conjunction with a recently developed cavitation index to predict the LOE of UCAs.

## Session 1aNS

## Noise and ASA Committee on Standards: Minimum Sound Requirements for Hybrid and Electric Vehicles to Protect Pedestrians

Brigitte Schulte-Fortkamp, Cochair
*Technische Universität Berlin, Einsteinufer 25, Berlin 10178, Germany*

William J. Murphy, Cochair
*Hearing Loss Prevention Team, Ctr. for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 4676 Columbia Parkway, M.S. C-27, Cincinnati, OH 45226-1998*

**Chair's Introduction—8:25**

### *Invited Papers*

**8:30**

**1aNS1. Optimizing detection of masked vehicles.** Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

The 250 million vehicles in the United States make so much noise that it is often difficult for pedestrians to hear a particular approaching vehicle, perhaps one that they do not see or realize is there. This masking effect is a pedestrian safety concern. Solving the general problem of masked vehicles is very difficult, because each vehicle plays conflicting roles–it needs to be heard, while it simultaneously masks other vehicles. This is especially true in urban settings where many vehicles and pedestrians are present. This paper seeks to identify the optimal vehicle noise level to minimize masking.

**8:50**

**1aNS2. Sound design concepts meeting minimum sound requirements—Advantages and disadvantages.** André Fiebig and Klaus Genuit (HEAD Acoust. GmbH, Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

A young field of activity for acoustic engineers has emerged in the context of vehicle exterior noise design considering the aspect of pedestrian safety. The fear of blind associations, public authorities, and governmental agencies to have a high collision risk for visually impaired persons with "quiet" vehicles led to the definition of new minimum sound requirements for hybrid and electric vehicles ostensibly leading to ten times fewer pedestrian and cyclist injuries. The requested sound should be detectable under a wide range of background noises when a vehicle is traveling under 18 mi/h. It is still open, whether hybrid and electric vehicles cause verifiably a higher collision risk for vulnerable persons and consequently the German Federal Environmental Agency rejects the idea of introducing additional sounds to enhance the audibility of vehicles operating in electric mode. Nevertheless acoustic engineers have to deal with the new requirements in the context of vehicle exterior noise and must find sustainable solutions considering product sound quality, detectability of vehicles and noise annoyance in equal measure. The paper will discuss the context-related need for additional sounds to improve pedestrian safety and the conceptual scope of sound design taking into account community noise aspects.

**9:10**

**1aNS3. Preliminary studies on the relation between the audio-visual cues' perception and the approaching speed of electric vehicles.** Luigi Maffei, Massimiliano Masullo, Maria Di Gabriele, and Francesco Sorrentino (Dept. Architecture and Industrial Design, Seconda Università di Napoli, Via S.Lorenzo, Aversa 81031, Italy, luigi.maffei@unina2.it)

For decades, "quiet" and "zero" emission vehicles have been considered the challenge for researchers and for the industry. Today, despite the great results obtained in the fields of air and noise pollution, the electric vehicles (EV) and hybrid vehicles (HV) have raised an important question regarding the pedestrian safety. At the speed permitted in urban areas (<50 km/h), these vehicles are considerably quieter than the traditional ones powered by gasoline or diesel. Nevertheless the amount of auditory cues associated to the approaching of these vehicles can be reduced, and this can determine an increase of the risk of accidents for the pedestrians. Even though the recent studies on this problem are focused, mainly, on the minimum sound levels and on the spectral content of the approaching vehicles, further aspects of the semantic content's change of the event should be considered. In this paper, a preliminary investigation on the relationship, and possible incoherence, among the approaching speed of the vehicles, the auditory cues, and the semantic content was performed, and results are presented. For the investigation, an immersive virtual reality environment was used.

9:30

**1aNS4. E-mobility as a constituent of the environment—Does it force a change of habits?** Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)



The future of car manufacturing is strongly led by the introduction of the e mobile cars considering the society's resources and its given economy with respect to acoustic ecology. Hence, no manufacturing questions will be raised, but the development of the e mobile cars related to the needs of the society's changes will be discussed. As sound design is always related to the status of a society, the acceptance of e mobile cars will be related to the way the society forces this new lifestyle product. Soundscape can be considered as a dynamic system characterized by the time-dependent occurrence of particular sound events embedded in specific environments. Soundscape describe ambiances and combine the daily recurrent patterns of sound multi-factorial in the process of analysis. An adequate evaluation of environmental noise will reflect the continually varying acoustical scenery and its specific perception in context. Therefore, the main message is focused on that any ambiance whether it is urban or rural will be a kind of composition where sounds play an informative role. The information is not based on the loudness level, it is based on the meaning of the sound and people's reaction to it.

**9:50–10:05 Break**

10:05

**1aNS5. What is the biggest threat to health and safety: Quiet or noisy vehicles?** Ulf Sandberg (Swedish National Rd. & Transport Res. Inst. (VTI), VTI, Linkoping SE-58195, Sweden, ulf.sandberg@vti.se)

It is assumed by US federal authorities that vehicles, driven in electric mode, either hybrid or pure electric vehicles, are so quiet that they are a safety hazard for pedestrians and cyclists in traffic. A regulation on this topic is already proposed in the United States, and in Europe, the Parliament has voted in favor of a similar regulation. ISO has worked out a measuring method for the "quietness" of vehicles; assuming that this can be used to require quiet vehicles to have some sound added. Based on earlier publications on this subject by the author, and reviewing some recent papers and reports, the author explores the assumed problem, and puts it in the perspective of traffic safety as well as the detrimental effects of noise on humans. Special concern is given to the relation between "quiet" and "noisy" vehicles. It is argued that it is far more justified to work in toward reducing the noise emission of the noisiest vehicles in traffic than adding noise to the quiet ones, as in any traffic situation, masking by other sounds is more important than the sound level per se, and masking is largely influenced by noisy trucks, busses, and motorcycles.

## *Contributed Papers*

10:25

**1aNS6. An innovative method for the sonification of quiet cars.** Sebastien Denjean, Vincent Roussarie (PSA Peugeot Citroen, route de Gisy, Velizy Villacoublay 78140, France, sebastien.denjean@mpsa.com), Solvi Ystad, and Richard Kronland Martinet (Laboratoire de Mécanique et d'Acoustique, Marseille, France)

With the development of electric motorizations, the acoustic feedback that the driver perceives in the passenger compartment has deeply changed. The vanishing of engine noise is often associated with better comfort, but it also represents a substantial loss of information for the driver. Internal combustion engine noise indeed plays a major role in the acoustic contribution of the multisensory perception of motion. In a previous experiment, we showed that it was more difficult for drivers to correctly estimate the vehicle speed without engine noise. Thus, more attention is needed from the driver to correctly regulate the speed in electric cars than in internal combustion engine cars. Consequently, we developed an innovative sonification method to compensate for the loss of auditory information in electric cars. The generated sounds are directly controlled by the vehicle's driving parameters, which give relevant cues on motion. This method has been tested in a driving simulator study to validate its suitability and to show that the control of sound properties can be adjusted to give precise information to the driver.

10:40

**1aNS7. The cart before the horse? National noise policy for hybrid and electric vehicles.** Dennis Weidemann (2633 Granite Rd., Fitchburg, WI 53711, dweid@mac.com) and Leslie D. Blomberg (Noise Pollution Clearinghouse, Montpelier, VT)

As the federal government proceeds to define noise policy for hybrid and electric vehicles, questions are being raised about the adequacy of information and science behind that policy. This paper identifies information required to produce effective noise policy for hybrid and electric vehicles, and examines federal policy with respect to those requirements. What do we need to know? What do we know? And most importantly, what don't we know?

10:55

**1aNS8. Sound quality evaluation of an automotive horn.** Hee Soo Kang and Sang Kwon Lee (Mech. Eng., Inha Univ., Incheon, South Korea, sangkwon@inha.ac.kr)

The sound quality of horn is important in views of two points. One is that it should be loud. The other one is that it is nice for a driver to hear the sound. Therefore, the exterior sound due to horn is related to loudness. The sound radiated from horn should be loud and the sound depends on the design of horn by itself. However, the interior sound due to horn depends on two paths. One is airborne path. The other one is structure-borne path. The sound quality of horn inside a car is related to the combination of sounds transferred throughout both paths. In order to evaluate the sound quality inside a car, the objective method evaluating the sound quality due to both paths is need to be developed. The paper presents the method evaluating objectively the sound quality of a horn inside a car based on the spectrum analysis and the octave analysis.

**11:10–11:45 Panel Discussion**

## Session 1aPA

# Physical Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics: Nonlinear Sound and Ultrasound Field Reconstruction and Related Applications I

Yong-Joe Kim, Cochair
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Je-Heon Han, Cochair
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**Chair's Introduction—8:55**

### *Invited Papers*

**9:00**

**1aPA1. Time reversal enabled elastic wave data communications using sensor arrays.** Yuanwei Jin (Eng. and Aviation Sci., Univ. of Maryland Eastern Shore, 30806 University Blvd. South, Princess Anne, MD 21853, yjin@umes.edu), Yujie Ying (Civil and Environ. Eng., Carnegie Mellon Univ., Pittsburgh, PA), and Deshuang Zhao (Inst. of Appl. Phys., Univ. of Electron. Sci. and Technol. of China, Chengdu, Sichuan, China)

The Data communication on pipe-like structures presents a unique opportunity for gas pipeline industry or oil drilling and exploration industry to meet the need for information transfer or data telemetry over a long distance. Typically, the operating environment under which the data telemetry is required for gas pipeline or oil drilling is not suitable for traditional communication modalities such as wireless radio communication, optical cable communication, or acoustic sound communication. Elastic waves are waves that can propagate over a long distance on solid medium such as metal pipes or strings. We have demonstrated experimentally the use of elastic wave for data communication over metal pipes in a single-input single-output configuration, which yields a data rate of 20 kb/s by time reversal pulse position modulation (TR-PPM). The result is obtained by exploring the reciprocity of elastic wave channel using time reversal without relying on additional processing techniques such as channel equalization or error control coding. In this paper, we report our experimental and theoretical investigation of TR-PPM scheme using sensor arrays to further improve the data communication throughput.

**9:20**

**1aPA2. Nonlinear ultrasound simulation for heterogeneous nonlinear coefficient media.** François Varray, Christian Cachard (CNRS UMR5220; Inserm U1044; INSA-Lyon; Université Lyon 1, Creatis, Université de Lyon, 7 av jean capelle, Villeurbanne 69621, France, francois.varray@creatis.insa-lyon.fr), Piero Tortoli (Dipartimento di Ingegneria dell'Informazione, Università di Firenze, Microelectron. Syst. Design Lab., Firenze, Italy), and Olvier Basset (CNRS UMR5220; Inserm U1044; INSA-Lyon; Université Lyon 1, Creatis, Université de Lyon, Villeurbanne, France)

In medical ultrasound, the use of simulation tools is very common to develop and test new imaging strategies. The existing tools are mainly limited to the linear propagation of ultrasound waves in tissue. A specific difficulty is due to the fact that the nonlinearity of investigated media can be different at different depths and the local value of the nonlinear coefficient must be taken into account to provide realistic simulated images. CREANUIS is a new two-step tool to simulate ultrasound images of linear and nonlinear media. It is based on a generalized angular spectrum method to consider the nonlinear forward propagation with heterogeneous nonlinear map and uses a reconstruction algorithm to create the corresponding radio frequency image. The simulated linear images have been validated through the comparison with images provided by FieldII, the reference tool in linear ultrasound simulation. CREANUIS allows simulating various nonlinear imaging techniques such as amplitude modulation or pulse inversion. Examples of images obtained simulating tissues with heterogeneous coefficients of nonlinearity will be presented. The comparison with experimental images provided by the ULA-OP ultrasound system and a grayscale phantom highlight the performance of CREANUIS images. CREANUIS is freely available at http://www.creatis.insa-lyon.fr/site/fr/CREANUIS.

**9:40**

**1aPA3. Transient, planar, nonlinear acoustical holography for reconstructing acoustic pressure and particle velocity fields.** Yaying Niu (Noise and Vib. Group, KBR, Inc., Houston, TX), Zheyu Zha, and Yong-Joe Kim (Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77843-3123, zhazheyu@tamu.edu)

A steady-state, nonlinear nearfield acoustical holography procedure, based on the Westervelt Wave Equation (WWE), was developed by the authors of this article to accurately reconstruct nonlinear acoustic "pressure" fields. Here, a "transient," nonlinear acoustic holography algorithm is introduced that can be used to reconstruct three-dimensional, nonlinear acoustic pressure as well as particle velocity fields from two-dimensional acoustic pressure data measured on a measurement plane. This procedure is based on the Kuznetsov Wave

Equation (KWE) that is directly solved by applying temporal and spatial Fourier Transforms to the KWE. When compared to the WWE-based procedure, the proposed procedure can be used to reconstruct acoustic particle velocity fields in addition to acoustic pressure fields. It can also be applied to multi-frequency source cases where each frequency component can contain both linear and nonlinear components. The KWE-based procedure is validated by conducting four numerical simulations with: (1) an infinite-size panel vibrating at a single frequency, (2) a pulsating sphere with a bifrequency excitation, (3) a finite-size, vibrating panel generating bended wave rays, and (4) an ultrasound transducer with a transient excitation. The numerical results show that holographically projected acoustic fields match well with directly calculated ones.



**10:00–10:15 Break**

**10:15**

**1aPA4. Reversible quasi-holographic line-scan and circular aperture processing for acoustic imaging and feature isolation.** Daniel Plotnick, Phillip L. Marston, David J. Zartman, and Anthony R. Smith (Washington State Univ., 1510 NW Turner DR, Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com)

One-dimensional line scans of objects by a collocated source/receiver can be processed via a linear quasi-holographic method [Baik *et al*., J. Acoustic Soc. Am. **130**, 3838–3851 (2011)] to produce an image in a way that is reversible for signal isolation. Distinct image features such as those due to edge diffraction, specular reflection, or elastic effects can then be extracted in the image domain. Images are then reverse processed to allow examination of the isolated features in time and spectral domains and removal of clutter [Zartman *et al*., *Proceedings of Meetings on Acoustics (POMA)* (June, 2013), Vol. 19, p. 055011]. In related research monostatic data from a circular synthetic aperture may be reversibly processed in a way favorable for feature extraction [Marston *et al*., *Proc. IEEE Oceans 2011* (2011)]. Experimental examples comparing extracted features with physical models will be discussed. [Work supported by ONR.]

## *Contributed Papers*

**10:35**

**1aPA5. Creation of coherent complex pressure measurements through overlapping scan-based measurements.** Jazmin S. Myres (Phys. and Astronomy, Brigham Young Univ., 2402 Timothy Cir, Pleasant Grove, UT 84062, jazmin.myres@gmail.com), Kent L. Gee, Tracianne B. Neilsen, and Alan T. Wall (Physics and Astronomy, Brigham Young Univ., Provo, UT)

In scan-based array measurements, stationary reference sensors are needed to temporally correlate the different measurement scans and produce coherent complex pressure fields. Because the number of references required increases with the number of subsources contributing to the sound field, an extended, partially correlated source (e.g., a turbulent jet) comprising many ill-defined sources can result in significantly increased measurement complexity. A different approach to creating spatiotemporally coherent pressures is demonstrated here. Military jet data have been used to explore "stitching" together a complex pressure field by spatially overlapping measurement scans instead of using separate reference channels. Various methods of stitching have been explored and the most robust method identified. Unwrapping of intrascan phases is first accomplished with a two-dimensional phase unwrapping algorithm. Individual scan positions are then stitched together using median phase differences between multiple adjacent scans to create coherent planes of data. Amplitude-stitching is done by averaging across scans and preserving the integrated squared pressure across the overall aperture. This method works well for low-frequency jet data, where there is not a ground-based interference null creating a physical phase discontinuity. This technique provides direction for efficient experimental design for scan-based array measurements of extended sources. [Work sponsored by ONR.]

**10:50**

**1aPA6. Evolution of the wave steepening factor for high-amplitude sound propagation.** Michael B. Muhlestein, Kent L. Gee, Tracianne B. Nielsen, and Derek C. Thomas (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, mimuhle@gmail.com)

One suggested metric to quantify nonlinear noise propagation is the wave steepening factor (WSF) [J. Gallagher, AIAA-82-0416 (1982)], defined as the absolute value of the mean negative slope of an entire waveform over the mean positive slope. Qualitatively, the WSF describes a waveform's average distortion, but a quantitative analysis has not been performed previously. To aid in the interpretation of the WSF, its evolution is presented in analytical fashion for the Earnshaw and Khokhlov solutions to the Burgers equation. Furthermore, the results of an analysis of the effects of finite sampling rates on the estimation of the wave steepening factor for these ideal cases are presented. A generalized Burgers equation-based model has been used to analyze the numerical evolution of the WSF for atmospheric and boundary layer absorption and dispersion and compare with plane wave tube experiments. From the numerical simulations, it is found that the WSF is dominated by initial level and frequency and is relatively less affected by absorption, bandwidth, and initial statistics for noise signals. Furthermore, the WSF is shown to be relatively insensitive to reduced sampling rates.

**11:05**

**1aPA7. Application of Mach-Zehnder interferometer to characterize spark-generated spherical N-waves in air.** Petr Yuldashev (Dept. of General Phys. and Condensed Matter Phys., Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Maria Karzova (Dept. of Acoust., Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Philippe Blanc-Benon, Sébastien Ollivier (LMFA UMR CNRS 5509, Ecole Centrale de Lyon, Université Lyon I, Université de Lyon, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, Ecully 69134 Ecully Cedex, France, Philippe.blanc-benon@ec-lyon.fr), and Vera Khokhlova (Dept. of Acoust., Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation)

Spark-generated spherical N-waves in air are measured by means of the Mach-Zehnder interferometer. A thin laser beam with sub-millimeter cross-section is used to obtain the highest time resolution. The light interference pattern is registered by a photodiode connected to a 10 MHz wideband amplifier. The inverse Abel transform is applied to the phase signal of the interferometer to reconstruct the pressure waveform. The peak positive pressure of 3000 Pa is obtained at the distance 10 cm from the source. It is shown that for the realized experimental setup a time resolution of 0.4 $\mu$s is reached. This resolution is 6 times better than the time resolution of condenser microphones (2.5 $\mu$s for the Brüel & Kjær 1/8 in.). The experimental waveforms measured at distances up to 100 cm are compared with the results of numerical modeling based on the generalized Burgers equation. The experimental waveform measured at 10 cm is set as a starter waveform for the numerical model. The theoretical waveforms are found to be in a good agreement with the measurements for all propagation distances with less than 5% difference for the peak positive pressure. [Work supported by PICS RFBR 10-02-91062/CNRS 5603, RFBR grant 12-02-31830-mol$_a$, the President of Russia MK-5895.2013.2 grants and by LabEx CeLyA ANR-10-LABX-60 and SIMMIC ANR-2010-BLAN-0905-03.]

# Session 1aUW

## Underwater Acoustics and Acoustical Oceanography: Deep Water Acoustics I

John Colosi, Cochair
*Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943*

Karim G. Sabra, Cochair
*Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405*

Kathleen E. Wage, Cochair
*George Mason Univ., 4400 University Dr., Fairfax, VA 22030*

### *Invited Papers*

**9:00**

**1aUW1. Deep water acoustic propagation in the northern Philippine Sea: Comparison of observed ray travel times with a non-acoustic state estimate.** Bruce Cornuelle, Ganesh Gopalakrishnan, Matthew Dzieciuch, and Peter Worcester (UCSD-Scripps Inst. of Oceanogr., 9500 Gilman Dr., Dept 0230, La Jolla, CA 92093-0230, bcornuelle@ucsd.edu)

Six transceivers and one distributed vertical line array (DVLA) receiver were moored in the northern Philippine Sea from April 2010 to May 2011 for low-frequency (325 Hz and below), deep water propagation experiments. The shortest range was about 130 km and the longest about 640 km. A 2009 equipment test yielded travel times between a moored source and a more limited DVLA at about 185 km range along one of the 2010–2011 paths. Experimental goals included understanding the impacts of fronts, eddies, and internal tides on acoustic propagation and determining the utility of acoustic measurements for ocean state estimation. This presentation will report on comparisons between ray travel times computed in a state estimate for the Northern Philippine Sea excluding the ray data, and observations from the 2009 and 2010–2011 experiments. The state estimate uses satellite sea surface height and sea surface temperature observations as well as Argo and XBT profiles. Ray arrival times calculated from the state estimate will be compared to observations as a simple estimate of the novel information present in the data.

**9:20**

**1aUW2. What acoustic travel-times tell us about the ocean.** Brian Powell (Oceanography, Univ. of Hawaii, 1000 Pope Rd., MSB, Honolulu, HI 96822, powellb@hawaii.edu), Bruce Cornuelle (Atmospheric Science & Phys. Oceangr., SIO, La Jolla, CA), and Colette Kerry (Oceanography, Univ. of Hawaii, Honolulu, HI)

Measurements of acoustic ray travel-times in the ocean provide synoptic integrals of the ocean state between source and receiver. It is known that the ray travel-time is sensitive to variations in the ocean at the transmission time, but the sensitivity of the travel-time to spatial variations in the ocean prior to the acoustic transmission have not been quantified. Using an advanced numerical model, we can identify both the ocean dynamics that control the ocean state along a ray path and quantify the informational content of the ray travel-time observation. This study examines the sensitivity of ray travel-time to the temporally and spatially evolving ocean state in the Philippine Sea over a one year experiment. The travel-times are found to be sensitive to the internal tide generation prior to the sample time and to advective effects that alter density along the ray path. Temporal nonlinearity of these sensitivities suggest that prior knowledge of the ocean state is necessary to exploit the travel-time observations. After assimilating the travel-time observations, the contribution of the travel-time information to our estimation of the ocean state is quantified and evaluated to further identify what the travel-time observation reveals about the ocean.

**9:40**

**1aUW3. Tomography and the global predictability of mode-1 internal tides.** Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, 1013 N.E. 40th St., Seattle, WA 98105-6698, dushaw@apl.washington.edu)

Twenty years ago, temporally coherent, mode-1 internal tides were detected in the central North Pacific using acoustic tomography. The spatial sampling of tomography is a natural filter for these internal waves. The signals were conjectured to have originated from the Hawaiian Ridge 2000 km to the south, and shortly thereafter Ray and Michum found radiation of internal tides from that undersea ridge using satellite altimetry. The two data types are complementary: tomography demonstrates astonishing temporal coherence, altimetry demonstrates astonishing spatial coherence. In the Atlantic, diurnal internal tides, resonantly trapped between Puerto Rico and their 30N turning latitude, were observed by AMODE tomography, while the semidiurnal variability consisted of a complicated interference pattern. While several conventional in situ observations indicated incoherent components to mode-1 internal-tide variability, both tomography (directly) and altimetry (indirectly) show the incoherent component is minimal. Resolution of mode-1 is challenging, but, without basis, oceanographers have been reluctant to accept the tomography measurements. A combined frequency and wavenumber tidal analysis of altimetry data has recently demonstrated that the amplitude and phase of mode-1 internal tides are predictable, except, perhaps, in more variable regions. Attenuation appears to be negligible, and waves coherent in time and space extend across ocean basins.

### 10:00–10:15 Break

## *Contributed Papers*



### 10:15

**1aUW4. Passive acoustic thermometry of the deep water sound channel using ambient noise.** Katherine F. Woolfe, Shane Lani, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr., Atlanta, GA 30332-0405, kfaist3@gatech.edu)

Cross-correlation processing of ocean ambient noise has been proposed as a totally passive alternative to existing active methods for sensing the ocean environment such as acoustic tomography or acoustic thermometry. To this end, we investigated the spatial coherence of low frequency (f $<$ 40 Hz) ocean noise recorded in the deep sound (SOFAR) channel to demonstrate the feasibility of passive acoustic thermometry. Continuous recordings of ambient noise obtained from hydroacoustic stations of the International Monitoring System were processed between the years 2006 and 2012. Each hydroacoustic station uses two triangular horizontal arrays separated by approximately 100 km, and each array has three hydrophones. Coherent arrivals were extracted from time-averaged cross-correlations between the two spatially separated triangular arrays. A beamforming procedure, using data-derived adaptive weights, was used to track the seasonal fluctuations of the arrival time of the coherent wavefronts throughout the 6 years. These measured travel-time fluctuations were estimated to primarily result from seasonal temperature fluctuations on the order of 0.1°C in the SOFAR channel, which is consistent with independent temperature measurements made by the Argo float global array in the vicinity of the hydroacoustic stations over the same time period.

### 10:30

**1aUW5. Seasonal dependence of ambient noise in the North Pacific.** Mehdi Farrokhrooz, Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., 4450 Rivanna River Way PMB3740, Fairfax, VA 22030, mfarrokh@masonlive.gmu.edu), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

One characteristic of deep ocean noise is the decrease in ambient levels below the conjugate depth (Urick, 1975). Measurements made by Gaul *et al.* (JOE, 2007) and Morris (JASA, 1978) in the North Pacific indicate that the slope of the noise below the conjugate depth varies with location. Gaul *et al.*'s measurements show a much steeper slope than those of Morris, and this difference can be attributed to bathymetric effects (Shooter, JOE, 1990). The experiments described by Gaul *et al.* and Morris took place in September 1975 and September 1973, respectively. Noise levels measured during SPICEX in September 2004 are comparable to those recorded by Morris 31 years earlier. While the earlier experiments took place over 2–11 day periods, the SPICEX arrays were deployed for a year. Since the conjugate depth varies seasonally, it is expected that the noise profile also varies seasonally. This talk investigates the seasonal variations of noise in the SPICEX data set, focusing on the 10–100 Hz band. The analysis includes a discussion of the effects of local wind.

### 10:45

**1aUW6. Toward on-the-fly ray identification for RAFOS-2 navigation.** Bruce Howe, Eva-Marie Nosal, and Lora J. Van Uffelen (Ocean and Resources Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 402, Honolulu, HI 96822, bhowe@hawaii.edu)

RAFOS-2 has been proposed as an undersea equivalent of GPS. By using broadband source signals and resolving the multipath arrival structure, long range positioning accuracy can be ~100 m as demonstrated by Van Uffelen *et al.*, J. Acoust. Soc. Am. **133**, 3444 (2013). Further improvement requires ray identification to be performed for each instantaneous position. Using the PhilSea10 experiment as a test case, we show that the range independent arrival pattern amplitude(range, depth, travel time; ray ID) can be simply parameterized. This may lead to more efficient algorithms for real time in situ navigation of mobile platforms. [Work funded by the Office of Naval Research.]

### 11:00

**1aUW7. Acoustic localization of seagliders in the Philippine Sea using broadband transmissions from moored acoustic sources.** Lora J. Van Uffelen, Bruce Howe, Eva-Marie Nosal (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 1000 Pope Rd., MSB 205, Honolulu, HI 96815, loravu@hawaii.edu), Glenn Carter (Oceanography, Univ. of Hawaii at Manoa, Honolulu, HI), Peter Worcester, and Matthew Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Seagliders obtain precise GPS positioning at surfacing events between dives, but can be underwater for several hours and travel several kilometers during a single dive, during which time less is known about the position of the glider. Four Seagliders deployed in the Philippine Sea from November 2010 to April 2011 were equipped with Acoustic Recorder Systems, enabling them to record over 2000 transmissions from five moored broadband acoustic sources at ranges up to 700 km and depths up to 1000 m. These sources sequentially transmitted linear FM sweeps from approximately 200 to 300 Hz at 9-min intervals. Source-receiver ranges, determined from travel-time offsets of the received pattern of acoustic arrival peaks, are combined for sets of source transmissions using the stochastic inverse method of least squares to estimate the position of the glider at the time of acoustic reception. Because the source transmissions are separated in time and because the glider is in constant motion, it can move appreciably between source receptions. A linear model of glider motion is incorporated into the least-squares analysis to account for this motion and to simultaneously estimate glider velocity. These acoustically derived estimates of glider position are compared with the positions estimated by a glider kinematic model.

### 11:15

**1aUW8. Seabed loss and dispersion analysis of broadband propagation data measured in deep water basin.** David P. Knobles and Jason D. Sagers (ARL, UT at Austin, 10000 Burnet Rd., Austin, TX 78758, knobles@arlut.utexas.edu)

Experimental acoustic data taken in a deep-water environment are analyzed for information on the intrinsic attenuation and volume scattering in the seabed. Sound Underwater Signaling (SUS) explosive sources were deployed in the Gulf of Oman basin with a water depth of about 3300 m. Received pressure time series were recorded on three hydrophones located at about 500, 1650, and 3200 m depth. The sound speed profile is downward refracting, and the seabed is a thick mud sediment with a sound speed ratio of less than unity. A positive sound speed depth gradient is the main mechanism that returns energy to the water column. The time series arrival structure for range scales of about 80 km is composed of a well-defined sequence of ray-like arrivals corresponding to an increasing number of bottom interactions with deeper turning points in the sediment. Time-frequency dispersion analyses of modeled and measured received time series in the 5–600 Hz band are applied in identifying the frequency dependence of loss mechanisms in the seabed. [Work supported by ONR.]

## Session 1pAA

# Architectural Acoustics and Signal Processing in Acoustics: Acoustics in Coupled Volume Systems

U. Peter Svensson, Cochair
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Ning Xiang, Cochair
*School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180*

## *Invited Papers*

**1:00**

**1pAA1. Free path statistics in coupled rooms.** Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany, mvo@akustik.rwth-aachen.de)

Statistics of free paths in room acoustics led to the statistical reverberation theory of diffuse fields in spaces and thus to the well-known reverberation formulas. The specific distribution of free paths, however, can also give a more specific insight into the sound field and its deviations from the expected result using the mean free paths. Recently, Hanyu (JASA **128**, 1140, 2010) extended the statistical reverberation theory toward a separation of absorption effects and mixing effects by scattering. This approach can predict nonlinear decay curves and the transition of energy between specular and scattered processes. In the work presented here, a concept of Vorländer (J. Sound Vib. **232**, 129, 2000) is revisited. Free path distributions are logged in the room simulation program RAVEN and displayed in real time with regard to the geometric configuration of coupled spaces, and to the influence of absorption and scattering at the boundaries. This allows the observation of the decay curves in dependence on the actual free path distribution. From this data, it is tried to further develop simple prediction scheme of decay curves in non-diffuse and coupled spaces.

**1:20**

**1pAA2. Inferring delays between subrooms in systems of coupled rooms.** Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com) and Jonathan Botts (Dept. of Media Technol., Aalto Univ., Espoo, Finland)

Delay-differential-equation models of reverberant-energy decay in systems of coupled rooms were recently introduced as a semi-empirical correction to standard statistical-acoustics models in order to better account for the delay of energy transfer between subrooms of a coupled system [Summers, J. Acoust. Soc. Am. **132**, EL129-EL134, 2012]. Here, a Bayesian approach to parameter estimation for these models is developed and evaluated on simulated and measured data. Using these data, the ability to invert for delays from measured decay curves is assessed and an interpretation of the physical meaning of those inferred delays is presented. Finally, the theoretical and mathematical relationships between higher-degree-of-freedom models resulting from the introduction of delay and those resulting from the introduction of additional decay rates to the standard sum-of-decaying-exponential model are considered, and the resulting implications for parameter estimation and model selection are described.

**1:40**

**1pAA3. Room acoustical conditions in coupled video conference rooms.** Erlend I. Gundersen (Aker Solutions, Lysaker, Norway) and U. Peter Svensson (Electron. and Telecommunications, Norwegian Univ. of Sci. and Technol., O.S. Bragstads pl. 2B, Trondheim NO-7491, Norway, svensson@iet.ntnu.no)

A video conferencing situation combines the acoustical properties of two rooms. The resulting convolution of the two room impulse responses leads to a total impulse response with a reverberation, which is not a classical exponential decay. As a consequence, relationships between parameters such as clarity and reverberation time will be significantly different from those in single rooms, and this will furthermore affect the combination's suitability for speech communication. In this study, a measurement survey is presented from 11 rooms with video conferencing equipment. Their volumes ranged from 24 to 117 $m^3$, and their mid-range reverberation times were between 0.29 s and 0.70 s. Median values were 82 $m^3$ and 0.41 s, respectively. Impulse responses were measured in all rooms and in a subsequent analysis stage, impulse response pairs were convolved, simulating a connection between the corresponding rooms. Those convolved IRs were analyzed in terms of clarity and reverberation time. Recommended parameter values for single rooms were used as guidelines to understand the qualities of convolved rooms.

**2:00**

**1pAA4. Auralization of virtual rooms in real rooms using multichannel loudspeaker reproduction.** Soenke Pelzer and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustr. 50, Aachen 52066, Germany, spe@akustik.rwth-aachen.de)

When playing auralizations including virtual room reverberation through loudspeaker-based reproduction systems, there is usually an interaction between the auralized virtual rooms with the real room acoustics of the listening environment. In case of a listening room which is not perfectly dry, it is investigated which criteria the listening room should fulfill to avoid considerable interference with the auralizations. In a further step, a computer room acoustics simulation is extended to account for the listening space by modifying the resulting room impulse responses, so that the final room-in-room situation matches best to the targeted virtual room acoustics. The presented technique is then applied in a multimodal immersive virtual display (CAVE-like environment) where room acoustics are not matter of choice due to restrictions of projection screen materials and placement.



**2:20–2:35 Break**

**2:35**

**1pAA5. Energy decay analysis in coupled volumes using an acoustic wave simulator.** Anish Chandak, Lakulish Antani (Impulsonic, Inc., 222 Old Fayetteville Rd., C101, Carrboro, NC 27510, achandak@impulsonic.com), and Dinesh Manocha (Dept. of Comput. Sci., Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

The acoustics in coupled volumes present various challenges. For example, the sound energy decay curve is double-sloped indicating that a single reverberation time cannot be assigned to coupled volumes. Such a behavior depends on various factors like the aperture sizes between the coupled volumes, absorption coefficients of each volume, etc. Recently, many studies using computer simulation have been performed to better understand the acoustic behavior of coupled volumes. None of these studies perform full 3D acoustic wave simulation on coupled volumes as they are regarded as computationally expensive and limited to very small spaces. We perform acoustic wave simulation using a relatively new wave solver called adaptive rectangular decomposition (ARD). ARD is more accurate than geometric acoustics techniques. It has been demonstrated to provide reliable simulation results through comparison with measurement data in indoor and outdoor scenes. Furthermore, it is practical and can handle large acoustic spaces on a single desktop. In this study, we analyze energy decay curves in coupled volumes using ARD. We perform numerical simulation for various coupled volume configurations such as varying aperture sizes, absorption coefficients, etc., and present their decay curves for analysis.

**2:55**

**1pAA6. Multiple-slope sound energy decay investigations in single space enclosures with specific geometrical and material attributes.** Zühre Sü Gül (Architecture; R&D, Middle East Tech. Univ.; MEZZO Studyo LTD, METU Technopolis KOSGEB-TEKMER No112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com), Ning Xiang (Graduate Program in Architectural Acoust., School of Architecture, Rensselaer Polytechnic Inst., Troy, NY), and Mehmet Caliskan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

Acoustical coupling until now has basically been studied to define the peculiar sound field within acoustically coupled enclosures in which multiple-slope energy decays can often be observed. The key concern of this study is to reveal the potential of multiple-slope energy decay formation in over-size single space structures with particular geometry and distribution of materials in different acoustical performance characteristics. Specifically, multiple dome superstructures, composed of one central dome supported by semi-domes and transitional elements, are selected for the case studies. The interpretation of the acquired data is carried in order to broaden the definition of “the coupled volume system” with an emphasis on invisible sources and apertures of acoustical coupling in a “single volume system.” The methodology of the research involves joint use of *in-situ* acoustical measurements, acoustical modeling/simulation methods, and computational analyses. Bayesian analysis approach is applied in quantifying multiple-slope decay parameters. Initial results for selected cases indicate double and triple slopes for field tests and even more slope natures for simulations at various frequency bands. Future work aims to elaborate the mechanism of multiple-slope decay occurrence by energy feedback and distribution analysis.

## *Contributed Papers*

**3:15**

**1pAA7. Binaural effects in convolved room impulse responses.** Ulrich Reiter and U. Peter Svensson (Electron. and Telecommunications, Norwegian Univ. of Sci. and Technol., O.S. Bragstads pl. 2B, Trondheim NO-7491, Norway, svensson@iet.ntnu.no)

In situations like video and teleconferencing, the acoustical properties of two rooms are involved via a convolution of the two room impulse responses. Reverberation in the two rooms will play quite different roles since the sender room reverberation will partially reach the listener, in the receiver room, from the source direction. Therefore, binaural suppression of reverberation will be less efficient for parts of the total impulse response. Here, the situation is analyzed in terms of a simple room impulse response model with a direct sound followed by an ideal exponential decay. Such a model permits parametric studies, even of convolved impulse responses. Listening test results will quantify the importance of these binaural effects for the perceived quality of speech signals.

**3:30**

**1pAA8. Stages with high ceilings, pipe organs, and active acoustics.** Roger W. Schwenke and Steve Ellison (Res. & Development, Meyer Sound Labs., 2832 San Pablo Ave., Berkeley, CA 94702, rogers@meyersound.com)

Two case studies are presented of rooms that have a high ceiling over the stage to accommodate a pipe organ. Svetlanov hall is the principal venue of Moscow’s International Performing Arts Center. It has two to four rows of chorus seating on two levels at stage left and right. The upstage wall is occupied by the largest pipe organ in Russia. It has a physical reverberation time of 1.7 s, which is within the accepted range for symphonic music, but longer reverberation times would be preferred for some pipe organ repertoire. Christopher Cohan Performing Arts Center at California Polytechnic State University has a proscenium and thrust stage. The Forbes Pipe Organ is housed on the stage right wall of the thrust stage. The symphony usually performs on the thrust stage entirely in front of the proscenium with the

solid decorative fire curtain down. Both rooms had poor communication on stage between performers, which led them to implement a solution using active acoustics. In both rooms all of the active acoustic elements overhead of the stage are on motors and can be retracted when not in use.

**3:45**

**1pAA9. Sound propagation to and around the balcony edge in a performance hall.** Liu Yee Cheung and Shiu Keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong, louisa.cheung@connect.polyu.hk)

To investigate the propagation of sound from the source to and around the balcony, over 150 points were measured in a 1420-seat auditorium. Radial points around the balcony edge with distances of 1.5 and 2 m, as well as intermediate points on the direct paths between the omni-directional sound source on the stage and the centers of the radial positions around the balcony edge, were also measured. A 1:20 scaled-model of this hall was done as a complement to the full-scaled measurement done to study the propagation in finer details. Various acoustic parameters were obtained. With the wave file of the impulse response recorded, the propagation pattern could be traced.

MONDAY AFTERNOON, 2 DECEMBER 2013 — UNION SQUARE 23/24, 1:00 P.M. TO 5:15 P.M.

## Session 1pAB

## Animal Bioacoustics: Signal Identification, Processing, and Analysis

Brian K. Branstetter, Chair

*National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106*

### *Contributed Papers*

**1:00**

**1pAB1. Acoustic feature extraction and classification in Ishmael.** David K. Mellinger (Coop. Inst.for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu)

Ishmael is a user-friendly bioacoustic analysis tool for Windows. It includes displays of sound waveforms and spectrograms, recording capability for real-time input, several methods for acoustic localization, beamforming, several methods for automatic call detection, and a sound annotation facility. Ishmael is intended for users wishing to analyze large volumes of data quickly and easily. Ishmael's capabilities for classification now include a feature-extraction module that implements noise-resistant acoustic characterizations based on Fristrup's AcouStat system (Fristrup, Woods Hole Tech. Rept. WHOI-92-04, 1992). These features can be used as input to classifier(s) connected to Ishmael and implemented in MATLAB. Examples of cetacean (odontocete and mysticete) sound feature extraction and classification will be shown.

**1:15**

**1pAB2. Can wavelets solve the cocktail party?** Mark Fischer (Aguasonic Acoust., P. O. Box 308, Rio Vista, CA 94571-0308, info@aguasonic.com)

There are many eco-systems that exist where avian populations are dense. In such environments, it is often the case that many different species may sing contemporaneously with significant overlap in the frequencies used by each. Work will be presented to support the hypothesis that wavelet processing allows an analog solution to this "cocktail party" problem.

**1:30**

**1pAB3. Identification of individual beaked whales in the northern Gulf of Mexico.** Juliette W. Ioup, George E. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148, jioup@uno.edu), Christopher O. Tiemann (Appl. Res. Labs., Univ. of Texas, Austin, TX), Natalia A. Sidorovskaia (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA), and Arslan M. Tashmukhambetov (Dept. of Physics, Univ. of New Orleans, New Orleans, LA)

Recent Littoral Acoustic Demonstration Center (LADC) multi-mooring Environmental Acoustic Recording System (EARS) data from the northern Gulf of Mexico are analyzed to deduce identifications of individual beaked whales. Procedures are built on previously applied self-organizing map techniques for clustering sperm whale clicks and beaked whale clicks from workshop data. Associating the clicks from beaked whales is difficult because, compared to sperm whales, beaked whales have lower source level, a narrower beam, and a faster rate of turning. Recordings of individual clicks can be clustered according to their time domain signal, frequency spectrum, or wavelet spectrum. For example, clicks clustered according to the magnitude of their frequency components show similarities for all the clicks in a class (representing an individual), but significant differences from class to class, suggesting that this approach has promise for identifying individuals. Recent work by Baggenstoss (J. Acoust. Soc. Am. **130**, 102–112 (2011); J. Acoust. Soc. Am. **133**, 4065–4076 (2013)), who has used cross correlations of clicks to assist in associating beaked whale clicks into trains, reinforces the idea that single click properties can be associated with individuals. [Research supported by SPAWAR and ONR.]

**1:45**

**1pAB4. Dolphin biosonar target detection in noise, auditory filter shape, and temporal integration.** Whitlow W. Au (Hawaii Inst. of Marine Biol., Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, wau@hawaii.edu)

The biosonar target detection in noise capability of the Atlantic bottlenose dolphin (*Tursiops truncatus*) has been studied as early as 1981 by Au and Penner. At that time, they presented results of the performance of two dolphins as a function of the signal energy to the density of the noise intensity (as in human psycho-acoustics) and not the signal energy to the noise energy ratio. Subsequently, two important pieces of information were determined, that is the auditory filter shape of the bottlenose dolphin in 2012 along with the temporal integration time for the reception of broadband biosonar echoes in 1988. In all but one experiment, it has been shown that the auditory system of odontocetes has a constant-Q characteristic. The biosonar detection capability of the bottlenose dolphin is revisited, and the target detection performance is now determined as a function of the received energy in the echo to the received noise energy. This presentation is an example of how some biosonar questions can only be answered after many years of related research and how sequences of research projects should be established so that current results can be applied to past results to gain a deeper understanding of the biosonar process.

2:00

**1pAB5. Sperm whale coda repertoires in the western Pacific Ocean.** Elizabeth L. Ferguson (Bio-Waves, Inc., 12544 Caminito Mira Del Mar, San Diego, CA 92130, eferguson@bio-waves.net), Thomas F. Norris, Cory A. Hom-Weaver, and Kerry J. Dunleavy (Bio-Waves, Inc., Encinitas, CA)

Sperm whales are social cetaceans that live in matrilineal family units, and inhabit all major ocean basins from the tropics to polar regions. They produce stereotyped patterns of 3 to 40 broadband clicks, termed "codas," that typically occur within a period of less than 3 s. Coda repertoires can be assigned to a "vocal clan," a type of social group used to define sperm whale population structure. Extensive studies of vocal clans have been conducted in the eastern tropical Pacific (ETP); however, little is known about sperm whale coda repertoires in the western Pacific. We reviewed codas recorded from independent sperm whale groups that were acoustically and visually encountered during two marine mammal surveys conducted in the Northern Mariana Islands (10 groups) and Palau region (3 groups), in 2007 and 2012, respectively. Three bioacousticians qualitatively classified codas to type, which indicated the presence of the "+1" and "regular" vocal clan. These data are now being analyzed using multivariate methods described by Rendell and Whitehead (2003) to quantitatively classify codas from each group. The identification of vocal clans within this region has implications for understanding the culturally linked stock distribution of sperm whales across the Pacific Ocean.

2:15

**1pAB6. The effects of site and instrument variability on recognizing odontocete species by their echolocation clicks.** Johanna Stinner-Sloan, Marie A. Roch (Dept. of Computer Science, San Diego State Univ., 5500 Campanile Dr, San Diego, CA 92182-7720, Johanna.RosalieChristina@gmail.com), and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA)

This work conducts a systematic study of the impact of variable site and instrument conditions on the performance of a species recognition task. Echolocation clicks were collected from six different sites in the Southern California Bight from multiple deployments of instruments with nine different preamplifiers. The classification performance of a Gaussian mixture model using cepstral features is examined on Risso's and Pacific White-Sided dolphins. One hundred three-fold Monte Carlo experiments are conducted. When grouped so that each acoustic encounter is either in the training or test set, a mean error rate of $1.9\%\pm4.4\sigma$ is obtained. In spite of correction for preamplifier response curves, grouping by preamplifier and site increases error dramatically to $20.9\%\pm18.1\sigma$ and $25.9\%\pm28.1\sigma$, respectively. We introduce noise compensation techniques that reduce error rates as follows: by encounter, $0.5\%\pm0.3\sigma$, by preamplifier, $1.7\%\pm2.3\sigma$, and by site, $9.4\%\pm16.7\sigma$.

2:30

**1pAB7. Using inter-click intervals to separate multiple odontocete click trains.** Eva-Marie Nosal (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Holmes Hall 405, Honolulu, HI 96822, nosal@hawaii.edu), Anders Høst-Madsen, and Jeremy Young (Elec. Eng., Univ. of Hawaii at Manoa, Honolulu, HI)

We developed a method to separate click trains from multiple odontocetes that relies on click timing only. The method assumes that inter-click intervals (ICIs) are slowly varying and that their distribution is known, though possibly with unknown parameters. ICIs are modeled as a renewal process and click trains are separated by maximizing the likelihood of click assignments based on ICI distributions. We present results from application of this timing-based separation method to simulated and real datasets. [Work supported by the National Science Foundation and the Office of Naval Research.]

2:45

**1pAB8. Comparisons of beaked whale signals recorded in eastern Atlantic to known beaked whales signals.** Odile Gerard (DGA, Avenue de la Tour Royale, Toulon 83000, France, odigea@gmail.com)

Beaked whales are a group of more than 20 genetically confirmed species; they are very elusive and were among the least known species until a few years ago. Because of their sensitivity to sonar, an increased research effort dedicated to these species and in particular to their signals started 10 years ago. Despite this effort, the signals of many beaked whale species remain unknown. Signals of Blainville's (*Mesoplodon densirostris*), Cuvier's (*Ziphius cavirostris*), Gervais' (*Mesoplodon europaeus*), and Northern Bottlenose Whales (*Hyperoodon ampullatus*) have been studied and described in different articles. The signals of three unknown species (different from the above mentioned species signals) have been reported. All seven species produce upsweep frequency modulated signals which seem species specific. In 2010, NATO Undersea Research Centre (NURC) conducted a sea-trial in Eastern Atlantic Ocean, Southwest of Portugal. Three different types of beaked whale signals were recorded. The characteristics of these signals will be presented and compared to the known signals of beaked whales. The conclusions of this analysis will be presented.



3:00–3:15 Break

3:15

**1pAB9. Whistle classification by spectrogram correlation.** Yang Lu and David Mellinger (Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, lu.yang@noaa.gov)

The spectral properties of whistles are investigated for sounds collected from six species of odontocetes in the North Pacific: bottlenose dolphins (*Tursiops truncatus*), melon-headed whales (*Peponocephala electra*), short- and long-beaked common dolphins (*Delphinus delphis* and *D. capensis*), and Hawaiian spinner dolphins (*Stenella longirostris longirostris*). The proposed method first uses the k-means algorithm to cluster whistles of each species into groups which represent their spectral contours. The features used for clustering are based on a method described by Mellinger *et al.* (J. Acoust. Soc. Am. **107**, 3518-3529). Short spectrogram kernels of different slopes are cross-correlated with training data to obtain histograms of slopes and other acoustic properties. The features are used in a random forest classification algorithm. The classified whistles which belong to different clusters but the same species are merged together, resulting in classification to species. The performance of these methods, as evaluated via confusion matrices, is compared in terms of accuracy and complexity with other extant algorithms.

3:30

**1pAB10. Robust recognition of whistle-like frequency contours by a bottlenose dolphin (*Tursiops truncatus*).** Brian K. Branstetter (National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, brian.branstetter@nmmpfoundation.org), Amy Black, Kimberly Bakhtiari, and Jennifer Trickey (G2 Software Systems Inc., San Diego, CA)

Whistle use by bottlenose dolphins serves several functions including individual identification, maintaining group cohesion, long-range communication, recruitment during feeding, and advertising emotional state. Whistles can vary considerably in amplitude, duration, and frequency. Despite this variability, dolphins must learn to associate specific whistle features (e.g., frequency contour) with meaningful events or objects, requiring plasticity in the dolphin's recognition abilities. To test robust recognition, a bottlenose dolphin learned to associate whistle-like frequency contours with arbitrary objects. For example, whistle-A would cue the dolphin to touch object-A, and whistle-B would cue her to touch object-B. The whistles were then altered by varying amplitude, duration, and transposing frequency. Changes in amplitude and duration had little effect on recognition, however frequency transposition as small as 1/3 octave resulted in poor performance. The results are discussed from a comparative cognition perspective.

**3:45**

**1pAB11. Sparse representation classification of dolphin whistles using local binary patterns.** Mahdi Esfahanian, Hanqi Zhuang, and Nurgun Erdol (Elec. Eng., Florida Atlantic Univ., 777 Glades Rd., EE-96 Bldg., Boca Raton, FL 33431, mesfahan@fau.edu)

A sparse representation classifier (SRC) has been adapted and applied to spectrograms to identify bottlenose dolphin whistles by their types. The classifier that relies on near completeness of the training features renders their choice no longer crucial as long as criteria are met to assure signal sparsity. Signal sparsity is ensured via the employment of a robust, effective, and computationally simple local binary patterns (LBP) operator that eliminates the need for costly denoising and contour tracking operations. The performance of the proposed method is compared to classifier-feature combinations of the K-nearest neighbor (KNN) and support vector machine (SVM) classifiers, and feature vectors of time-frequency contour parameters, Fourier descriptors, and raw data. The experimental results demonstrate superior accuracy and robustness of the proposed method to classify dolphin whistles into distinct call types. The method can be generalized to all narrowband signals with time varying spectra.

**4:00**

**1pAB12. Discrimination of baleen whales frequency-modulated downsweep calls with overlapping frequencies.** Hui Ou, Whitlow Au (Hawaii Inst. of Marine Biol., Univ. of Hawaii, Kaneohe, HI, wau@hawaii.edu), Sofie V. Parijs (Northeast Fisheries Sci. Ctr., National Marine Fisheries Sci., Woods Hole, MA), Erin M. Oleson (Pacific Fisheries Sci. Ctr., National Marine Fisheries Sci., Honolulu, HI), and Shannon Rankin (Southwest Fisheries Sci. Ctr., National Marine Fisheries Sci., La Jolla, CA)

Spectrograms generated with the pseudo Wigner-Ville distribution (PWVD) provide much higher simultaneous time-frequency (TF) resolution compared with the traditional method using the short time Fourier transform (STFT). The WV-type spectrogram allows bioacousticians to study the fine TF structures of the sound, such as the instantaneous frequency, instantaneous bandwidth, contour slope, etc. These features set the foundation of identifying sounds that are usually considered difficult to discriminate using the traditional method. However, the PWVD requires much higher computational effort than the STFT method. In this research, the advantage of the WV spectrogram analysis was demonstrated by a case study on frequency-modulated, downsweep sounds from fin whales, sei whales, and blue whales D-calls. These calls overlapped in frequency range and have similar time duration. Automatic detection of fin, sei or blue whales FM downsweeps using the traditional spectrogram methodology tend to be ineffective because of the large temporal ambiguities needed to achieve the necessary frequency resolution. However, their WV spectrograms showed distinguishable characteristics, for example, the TF contour of fin and sei whales exhibited concave and convex shapes respectively. A support vector machine (SVM) classifier was trained and tested based on the parameters extracted from the WV spectrograms.

**4:15**

**1pAB13. Fin whales (*Balaenoptera physalus*) in British Columbia sing a consistent song.** Barbara Koot (Dept. of Zoology, Univ. of Br. Columbia, Rm. 247, 2202 Main Mall, Vancouver, BC V6T 1Z4, Canada, b.koot@fisheries.ubc.ca), John K. Ford (Fisheries and Oceans Canada, Nanaimo, BC, Canada), David Hannay (JASCO Appl. Sci., Victoria, BC, Canada), and Andrew W. Trites (Dept. of Zoology, Univ. of Br. Columbia, Vancouver, BC, Canada)

Geographic differences in fin whale song that may be related to population structure have been documented in the Atlantic and Southern Oceans. However, information on the songs and population structure of fin whales in the North Pacific is limited. We analyzed fin whale songs recorded over 9 months by an Autonomous Underwater Recorder for Acoustic Listening device (AURAL, Multi-Electronique Inc.) deployed west of Vancouver Island, British Columbia (July 2010 to March 2011). Our analysis focused on internote intervals—the song characteristic that others have shown to display the most geographic variation. We found that beginning in mid-August and continuing to the end of our study, fin whales produced one stereotyped song consisting of alternating classic (C) and backbeat (B) notes. Internote interval evolved slightly over this time period, and small differences in interval length occurred among individual songs. However, all songs shared the same note arrangement (i.e., the interval between C and B notes was always 30% longer than the interval between B and C notes). All whales recorded in this study produced a similar song, suggesting that they may belong to the same acoustic population. Future studies will help to determine the spatio-temporal boundaries of this acoustic population.

**4:30**

**1pAB14. Vocal repertoire of Southeast Alaskan humpback whales (*Megaptera novaeangliae*).** Michelle Fournet (College of Earth Ocean and Atmospheric Sci., Oregon State Univ., 425 SE Bridgeway Ave., Corvallis, OR 97333, mbellalady@gmail.com) and Andy Szabo (Alaska Whale Foundation, Seattle, WA)

Humpback whales (*Megaptera novaeangliae*) are vocal baleen whales that exhibit complex social interactions across broad spatial and temporal scales. On low latitude breeding grounds, humpback whales produce complex and highly stereotyped "songs" as well as a range of "social sounds" associated with breeding behaviors. While on their Southeast Alaskan foraging grounds, humpback whales produce vocalizations during cooperative foraging events as well as a range of unclassified vocalizations for which the social context remains unknown. This study investigates the vocal repertoire of Southeast Alaskan humpback whales from a sample of 366 vocalizations collected over a three-month period on foraging grounds in Frederick Sound, Southeast Alaska. We used a two-part classification system, which included aural-spectrogram and statistical cluster analyses, to describe and classify vocalizations. Vocalizations were classified into 19 individual call types nested within four call classes. The vocal repertoire of Southeast Alaskan humpbacks shows moderate overlap with vocalizations recorded in Atlantic foraging grounds and along the Australian migratory corridor.

**4:45**

**1pAB15. The acoustic signature of the male northern elephant seal: Individual variation supports recognition during competitive interactions.** Caroline Casey (Ecology and Evolutionary Biol., Univ. of California Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, cbcasey@ucsc.edu), Colleen Reichmuth (Inst. of Marine Sci., UC Santa Cruz, Santa Cruz, CA), Selene Fregosi (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), Isabelle Charrier (Equipe Communications Acoustique, Université Paris Sud, Orsay, France), and Nicolas Mathevon (Laboratoire de Biologie Animale, Université Jean Monnet, Saint-Etienne, France)

Northern elephant seals (*Mirounga angustirostris*) have a polygynous breeding system in which adult males establish dominance hierarchies that determine access to females. Acoustic signaling plays an important role in settling fights between males, as stereotyped displays elicit appropriate behavioral responses from individuals without contact during an energetically demanding breeding season. To determine whether reliable differences exist in the acoustic displays of individuals and whether these differences function to convey identity, we behaviorally and acoustically sampled male seals during the breeding season. Vocalizations were recorded during competitive interactions and analyzed for spectral, temporal, and amplitude characteristics. A cross-validated discriminant function analysis revealed small differences within—and significant differences between—the calls produced by 17 adult males of known dominance status. To determine whether acoustic displays serve as individual signatures that males learn to recognize during the breeding season, we conducted playback experiments to test if having prior experience with a particular caller would influence the approach or avoidance response of the listener. Our findings reveal that these unique acoustic signals serve as individual vocal signatures, and males likely remember the identity of their rivals based on call features that have been associated with the outcome of previous competitive interactions.

5:00

**1pAB16. Detection of complex sounds in quiet and masked conditions by a California sea lion (*Zalophus californianus*) and a harbor seal (*Phoca vitulina*).** Kane A. Cunningham (Ocean Sci., Univ. of California at Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, kacunningham413@yahoo.com), Brandon Southall (Southall Environ. Assoc., Aptos, CA), and Colleen Reichmuth (Inst. of Marine Sci., Long Marine Lab., Univ. of California at Santa Cruz, Santa Cruz, CA)

Standard audiometric data, such as absolute detection thresholds and critical ratios, are often used to inform noise-exposure limits for marine mammals. However, these data are traditionally generated using simple stimuli, such as pure-tones and flat-spectrum noise, while natural sounds tend to have more complex structure. In this experiment, detection thresholds for complex stimuli were obtained in (a) quiet and (b) masked conditions for one California sea lion and one harbor seal. For part (a), three stimuli types were synthesized, each isolating a common feature of marine mammal vocalizations: amplitude modulation (AM), frequency modulation (FM), and harmonic structure. Detection thresholds in quiet conditions were then obtained for these stimuli at frequencies spanning the functional hearing range. For part (b), the same complex signals were combined with flat-spectrum noise or shipping noise. To test how well standard hearing data predict detection of complex sounds, the results of parts (a) and (b) were compared to *a priori* predictions based on previously obtained audiogram and critical ratio data. Preliminary results indicate that absolute detection thresholds for AM and FM stimuli are reliably predicted by audiogram data, but that thresholds for harmonic stimuli are lower than predicted, in some cases by more than 10 dB.



MONDAY AFTERNOON, 2 DECEMBER 2013 PLAZA A, 1:30 P.M. TO 5:00 P.M.

## Session 1pAO

## Acoustical Oceanography: Contributed Papers in Acoustical Oceanography (Poster Session)

Timothy K. Stanton, Chair
*Woods Hole Oceanogr. Inst., M.S. #11, Woods Hole, MA 02543-1053*

### *Contributed Papers*

**1pAO1. Observations of region-specific fish behavior using long- and short-range broadband (1.5—6 + kHz) active acoustic systems.** Timothy K. Stanton (Dept. Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., M.S. #11, Woods Hole, MA 02543, tstanton@whoi.edu), J. Michael Jech (Northeast Fisheries Sci. Ctr., NOAA, Woods Hole, MA), Roger C. Gauss (Acoust. Div., Code 7164, Naval Res. Lab., Washington, DC), Benjamin A. Jones (Oceanogr. Dept., Naval Postgrad. School, Monterey, CA), Cynthia J. Sellers (Dept. Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), and Joseph M. Fialkowski (Acoust. Div., Code 7164, Naval Res. Lab., Washington, DC)

Two broadband active acoustic systems, in concert with traditional narrowband systems and nets, were used to study distributions of fish in three regions within the Gulf of Maine. The long-range multi-beam broadband system detected fish out to 15 km range and the downward-looking short-range broadband system detected fish throughout the water column close behind the ship. The multi-year (2007–2011) study revealed distinct spatial patterns of fish and corresponding echo statistics in each region—diffusely distributed, sparsely distributed compact patches, and long (continuous) shoals. The broadband capabilities of the sonar systems (each spanning 1.5—6 + kHz) uniquely allow observations of resonance phenomena of the local swimbladder-bearing fish. The observed resonances were consistent with the fish species, sizes, and depths that were concurrently sampled in each area from a second research vessel. Spectral peak analysis also interestingly revealed the presence of distinct modes, which may be useful indicators of mixed-species and/or mixed-sized (e.g., juvenile and adult) assemblages of fish. [Work supported by Office of Naval Research.]

**1pAO2. The history detectives: Establishing the parameters of the 1960 Perth-Bermuda antipodal acoustic propagation experiment.** Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, 1013 N.E. 40th St., Seattle, WA 98105-6698, dushaw@apl.washington.edu)

In 1960 three 300-lb explosive shots were detonated off Perth, Australia at 3 am, 22 March (local) by HMAS Diamantina to determine if those sound signals could propagate the antipodal distance to the Bermuda SOFAR station. These data offer a rare measure of the ocean temperature a half century ago, averaged across large stretches of the Southern, South Atlantic, and North Atlantic Oceans. The accuracy of these data are determined by the accuracy of the essential parameters of the experiment, e.g., the time and position of the shots. The narrative of HMAS Diamantina the night of 21 March 1960 was reconstructed from the ship's log, the captain's Report of Monthly Proceedings, and other information. The experiment was conducted with care to obtain a precise measurement, subject to the resources available to the ship at the time. The largest uncertainty is in the position of the shots, determined by triangulation from shore landmarks in the evening, celestial navigation at dawn, and dead reckoning in between. In addition, the depth was measured at the time of the shots. The 1960 position was measured to an equivalent travel-time accuracy of about 3 s, biased toward closing the range to Bermuda.

**1pAO3. Fluctuations of the sound field in the presence of internal Kelvin waves in a stratified lake.** Boris Katsnelson (Marine Geosci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru), Andrey Lunkov (Wave Res. Ctr., General Phys. Inst., Moscow, Russian Federation), and Ilia Ostrovsky (Kinneret Limnological Lab., Oceanogr. Limnological Res., Haifa, Israel)

In stratified lakes internal waves has great ecological significance since they affect mixing, resuspension, material transport, chemical regime and ecosystem productivity. Reconstruction of spatio-temporal heterogeneity of the basin scale internal waves and their accurate parameterization are important tasks. The effect of internal Kelvin waves (IKWs) on spatiotemporal variability of the mid-frequency (1 kHz) sound field in a deep lake using geoacoustic modeling is studied. It is demonstrated that IKWs cause significant fluctuations of the sound field, such as horizontal shift of interference structure. This shift can be easily measured in situ and used for practical reconstruction of IKW parameters. Overall, it is suggested implementing the low-cost geoacoustic methodology for accurate parameterization of the basin scale internal waves and studying their dynamics.

**1pAO4. Measurements of diel variation of acoustic backscatter power from phytoplankton.** Hansoo Kim, Tae-Hoon Bok, Juho Kim, Dong-Guk Paeng (Dept. of Ocean Syst. Eng., Jeju National Univ.,102 Jejudaehakno, Jeju-si, Jeju Special Self-Governing Province 690-756, South Korea, hansoo5714@naver.com), Md Mahfuzur Rahman Shah, and Joon-Baek Lee (Dept. of Earth and Marine Sci., Jeju National Univ., Jeju, South Korea)

Phytoplankton is a primary producer in the ocean, and its vegetative activity plays an important role in controlling the global environment. In this study, the high-frequency acoustic signals were collected to evaluate the photosynthetic activity of phytoplankton during a day. The integrated backscatter power (IBP) from a dinoflagellate *Cochlodinium polykrikoides* was measured by a 5 MHz acoustic system during a 5-day cultivation period with a 14 h:10 h light:dark cycle. IBP increased by 0.6 dB in five days, but varied by 0.83 $\pm$ 0.1 dB during an irradiance cycle. The daily increase in IBP was a result of an increase in the number of cells during cultivation, while the diel variation was partly resulted from the variation of volume of cells by photosynthesis. In addition, cell division and separation might affect IBP. IBP of another dinoflagellate *Amphidinium carterae* Hulburt was also measured by a 10MHz transducer during a 3-day cultivation period while cell volume and photosynthetic capacity were measured four times a day (07:00, 12:00, 19:00, 24:00). This study suggests that high-frequency acoustics may be a meaningful tool to investigate the photosynthetic metabolism of a phytoplankton cell. [This work was supported by Defense Acquisition Program Administration and Agency for Defense Development under the contract UD130007DD.]

**1pAO5. Broadband normal mode energy metrics in the presence of internal waves.** Georges Dossot, Steven Crocker (NUWC, 1176 Howell Str., Bldg. 1320, Code 1524, Rm. 260, Newport, RI 02841, georges.dossot@navy.mil), James H. Miller, Gopu R. Potty (Ocean Engineering, Univ. of Rhode Island, Narragansett, RI), and Mohsen Badiey (College of Marine Studies, Univ. of Delaware, Newark, DE)

During the Shallow Water 2006 experiment a J-15 acoustic source deployed from the Research Vessel Sharp transmitted broadband (100–500 Hz) chirp signals 15 km away from a vertical line array. The array was intentionally positioned near the shelf-break front and in an area where internal waves are known to occur. Normal mode decomposition helps provide clues regarding the physics behind signal fluctuations due to internal waves, but often analyses are accomplished at single frequencies. A broadband modal beamformer approach is offered that extracts separate modal arrivals. A method for a mode-dependent matched filter is suggested which helps extract the separate arrivals in a low-signal environment. These methods can be used to compute mode-independent energy statistics which help explain signal fluctuations as an internal wave traverses the source-receiver path. [Work sponsored by the Office of Naval Research.]

**1pAO6. Acoustic mapping of ocean currents using networked distributed sensors.** Chen-Fen Huang (Inst. of Oceanogr., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, chenfen@ntu.edu.tw), TsihC Yang, and Jin-Yuan Liu (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan)

Distributed underwater sensors are expected to provide oceanographic monitoring over large areas. As fabrication technology advances, low cost sensors will be available for many uses. The sensors communicate to each other and are networked using acoustic communications. This paper first studies the performance of such systems for current measurements using tomographic inversion approaches to compare with that of a conventional system which distributes the sensors on the periphery of the area of interest. It then proposes two simple signal processing methods for ocean current mapping (using distributed networked sensors) aimed at realtime in-buoy processing, Tomographic inversion generally requires solving a challenging high dimensional inverse problem, involving substantial computations. Given distributed sensors, currents can be constructed locally based on data from neighboring sensors. It is shown using simulated data that results obtained using distributed processing are similar to those obtained from conventional tomographic approaches. The advantage for distributed systems is that by increasing the number of nodes, one gains a much more improved performance. Furthermore, distributed systems use much less energy than conventional tomographic system for the same area coverage. Experimental data from an acoustic communication and networking experiment are used to demonstrate the feasibility of acoustic current mapping.

**1pAO7. Low frequency scattering from fish schools: A comparison of two models.** Maria P. Raveau (Hydraulic and Environ. Eng., Pontificia Universidad Catolica de Chile, Vicuña Mackenna 4860, Macul, Santiago 7820436, Chile, mpraveau@uc.cl) and Christopher Feuillade (Phys. Dept., Pontificia Universidad Catolica de Chile, Santiago, Chile)

A theoretical comparison between two scattering models for fish schools was performed. The effective medium approach, based on the work of Foldy [Phys. Rev. **67**, 107–109 (1945)], which has previously been used to describe scattering from bubble clouds, was used to calculate the variations of the scattering length of a fish school with frequency, and with azimuth. Calculations were also performed with a model used previously to study collective back scattering from schools of swim bladder fish, which incorporates both multiple scattering effects between fish, and coherent interactions of their individual scattered fields [J. Acoust. Soc. Am., **99**(1), 196–208 (1996)]. Two different packing algorithms were used, in order to investigate the influence of the spatial distribution of fish on the scattering response of the school. Comparison of the two models shows good agreement in the forward scattering direction, where no frequency dependent interference effects are observed. The models indicate divergent results in the back scattering direction, where the arrangement of fish in the school strongly affects the scattering amplitude. The upper frequency limit of the effective medium approach is also discussed, and the effect of the depth of the school in the water column. [Research supported by ONR.]

**1pAO8. Sound absorption by fish schools: Forward scattering theory and data analysis.** Christopher Feuillade (Facultad de Física, Pontificia Universidad Católica de Chile, Av. Vicuña Mackenna 4860, Santiago 8920007, Chile, chris.feuillade@gmail.com) and María P. Raveau (Facultad de Ingeniería, Pontificia Universidad Católica de Chile, Santiago, Chile)

A model used previously to study collective back scattering from fish schools [J. Acoust. Soc. Am., **99**(1), 196–208 (1996)] is used to analyze the forward scattering properties of these objects. There is an essential physical difference between back and forward scattering from fish schools. Strong frequency dependent interference effects, which affect the back scattered field amplitude, are absent in the forward scattering case. This is critically important when analyzing data from fish schools to study their size, the species and abundance of fish, and fish behavior. Transmission data can be processed to determine the extinction of the field by a school. The extinction of sound depends on the forward scattering characteristics of the school, and inversion of absorption data to provide information about the fish should be based upon a forward scattering paradigm. Results are presented of an analysis of transmission data obtained during the Modal Lion experiment in September 1995, and reported by Diachok [J. Acoust. Soc. Am., **105**(4), 2107–2128 (1999)]. The analysis shows that using forward scattering typically leads to significantly larger estimates of fish abundance than previous analysis based on back scattering approaches. [Research supported by ONR.]

**1pAO9. Synthetic aperture geoacoustic inversion in the presence of radial acceleration dynamics.** Bien Aik Tan, Peter Gerstoft, Caglar Yardim, and William Hodgkiss (Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, btan@ucsd.edu)

Traditionally matched-field geoacoustic inversion experiments sampled the acoustic field on long arrays and require powerful transmissions in order to reduce parameter uncertainty. However, single-hydrophone based geoacoustic inversion methods are attractive compared to the ones using long arrays. A low signal to noise ratio (SNR), single-receiver, broadband, and frequency coherent matched-field inversion that exploits coherently repeated transmissions to improve estimation of the geoacoustic parameters have been previously proposed. The long observation time creates a synthetic aperture due to relative source-receiver motion. This paper extends broadband synthetic aperture geoacoustic inversion to cases where the velocity of the source/receiver changes. In addition, a pulse-by-pulse coherent processing using the Bayesian approach is proposed. The method is

demonstrated with low SNR, 100–900 Hz LFM data from the Shallow Water 2006 experiment.

**1pAO10. Intensity statistics for dual-band, 500 km, ocean acoustic transmissions in the Philippine Sea 2010 experiment.** Andrew A. Ganse, Rex K. Andrew, James A. Mercer (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aganse@apl.washington.edu), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The May 2010 component of the North Pacific Acoustic Laboratory's Philippine Sea experiment included low-frequency transmissions over a 500 km path from the APL-UW ship-suspended multi-port source to a distributed vertical line array (DVLA) with 149 working hydrophones spaced over most of the water column. The multi-port source has two acoustic resonances at approximately 200 and 300 Hz. To accommodate those resonances in this experiment, two M-sequence signals at those center frequencies were transmitted simultaneously. Data recorded over a 60 h data window in the experiment show deep fades in both frequency bands, i.e., frequency-dependent variations of 10–20 dB in acoustic arrival intensities on the DVLA. The measurements are processed with Wiener filtering based on source-monitoring receptions of the transmission on a hydrophone 20 m from the source, to compensate for effects of the power amplifier and the transducer. Statistics of the received acoustic intensities are reported for both frequency bands. Correlations in the intensity between these broadly separated frequency bands allow one to explore questions of the role in arrival-splitting (micro-multipathing) in the deep fades observed in this experiment. [Work supported by ONR.]



MONDAY AFTERNOON, 2 DECEMBER 2013 — GOLDEN GATE 2/3, 1:00 P.M. TO 3:45 P.M.

# Session 1pBA

## Biomedical Acoustics: Bubble Detection in Diagnostic and Therapeutic Applications II

Charles C. Church, Chair
*National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677*

### *Contributed Papers*

**1:00**

**1pBA1. Nonlinear dynamics and control of ultrasound contrast agent microbubbles.** James M. Carroll, Leal K. Lauderbaugh, and Michael L. Calvisi (Department of Mechanical and Aerospace Engineering, Univ. of Colorado, Colorado Springs, 1420 Austin Bluffs Parkway, Colorado Springs, CO 80918, mcalvisi@uccs.edu)

The nonlinear response of ultrasound contrast agent microbubbles is well-known and is important for both diagnostic and therapeutic purposes. A model of thin-shelled, spherical contrast agents subject to ultrasound is presented based on a modified form of the Rayleigh-Plesset equation combined with terms to account for the shell influence. Techniques from dynamical systems theory are employed to analyze the model and elucidate the complex behavior of contrast agents forced by ultrasound. It is shown that the contrast agent can undergo a sequence of bifurcations as the acoustic pressure amplitude is increased, leading to a transition from periodic to chaotic oscillations at sufficiently large forcing. Yet even at high forcing, regions of periodic behavior exist. The response of the contrast agent, however, is strongly dependent on the shell properties and the applied acoustic forcing. Furthermore, a nonlinear, sliding mode controller is developed and applied to the spherical contrast agent model to demonstrate the feasibility of using the nonlinear controller to modulate the contrast agent response through the incident ultrasound. Applications of the nonlinear control system to contrast agents include radius stabilization in the presence of an acoustic wave, excitation of radial growth and subsequent collapse, and generation of periodic radial oscillations while a contrast agent is within an acoustic forcing regime known to cause a chaotic response.

**1:15**

**1pBA2. A numerical model for large-amplitude spherical bubble dynamics in tissue.** Matthew T. Warnez and Eric Johnsen (Mech. Eng., Univ. of Michigan, 405 N Thayer, Ann Arbor, MI 48104, mwarnez@umich.edu)

In a variety of therapeutic and diagnostic ultrasound procedures (e.g., histotripsy, lithotripsy, and contrast-enhanced ultrasound), cavitation occurs in soft tissue, which behaves in a viscoelastic fashion. While stable bubble oscillations may occur in ultrasound, the most dramatic outcomes (tissue ablation, bleeding, etc.) are usually produced by inertial cavitation. Historically, Rayleigh-Plesset equations have been used to investigate the dynamics of spherical bubbles, including in biomedical applications. For large-amplitude bubble oscillations in tissue, it is clear that compressibility, heat transfer and nonlinear viscoelasticity play important roles. However, no existing model includes all of these effects. To address this need, we use a compressible Rayleigh-Plesset equation (Keller-Miksis) adjoined with heat conduction in conjunction with an upper-convected Zener viscoelastic model, which accounts for relaxation, elasticity, and viscosity. The partial differential equations describing the stress tensor components in the surrounding medium are solved using a spectral collocation method. The method proves to be robust even for strong bubble collapse. Numerical comparisons with previous models are made, comparisons to experiments are included, and the dependence of bubble dynamics on viscoelastic parameters is explored. This model is used to revisit the inertial cavitation threshold in biomedical settings.

**1:30**

**1pBA3. Delay differential equations for single-bubble dynamics in a compressible liquid.** Derek C. Thomas (Dept. of Phys. and Astronomy, Brigham Young Univ., N223 ESC, Provo, UT 84602, derekcthomas@gmail.com), Yurii A. Ilinskii (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Most common methods to include the effects of liquid compressibility in models for single-bubble dynamics rely on series expansions to some order in the inverse of the sound speed in the liquid. It has been shown that the Keller-Miksis model for single-bubble dynamics can be obtained from a series expansion of a delay differential equation related to the Rayleigh-Plesset equation for a single bubble. The iterative approach used to obtain the series expansion of the delay-based model becomes unworkable for more complicated models of bubble dynamics. Therefore, to provide an alternative, simpler method to model the effects of liquid compressibility, the

delay differential equation model proposed by Ilinskii and Zabolotskaya [J. Acoust. Soc. Am. **92**, 2837 (1992)] is analyzed directly. The results of the delay-based formulations are compared to those produced by models based on common series expansions. Alternative formulations of the delay differential equation are also considered and compared.

**1:45**

**1pBA4. Shell material parameter measurements of polymer ultrasound contrast agents.** Pavlos Anastasiadis, John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), Parag V. Chitnis, and Jeffrey A. Ketterling (Riverside Res., New York, NY)

Polymer shelled ultrasound contrast agents have been used in ultrasound imaging and tissue perfusion studies. The destruction of the agent produced by the rupture of the shell often through complicated buckling has been quantified with overpressure experiments and optical visualization. Approximate material parameters have been correspondingly estimated with this methodology but additional steps are needed to translate for a viable high throughput technique. Ultra-high frequency acoustic microscopy (1 GHz) provides a non-invasive method to image and measure the shell's elastic properties. Scanning acoustic microscopy at 1 GHz is used to determine the shell density and elastic modulus for polymer shelled agents of three different shell thicknesses. The effect bending thickness is estimated for comparison with overpressure experiments.

**2:00**

**1pBA5. Subharmonic response and threshold of polymer ultrasound contrast agents.** RIntaro Hayashi, John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), Jonathan Mamou, Parag V. Chitnis, and Jeffrey A. Ketterling (Riverside Res., New York, NY)

Analytical expressions for subharmonic threshold previously developed for free gas bubbles often been generalized and used for ultrasound contrast agents. However, many of those formulations were developed under the assumptions of mono-frequency steady state, continuous acoustic forcing which are typically not applicable to diagnostic and therapeutic ultrasound. Nonstationary subharmonic responses are investigated by analytical and numerical methods. The subharmonic threshold is investigated as means to differentiate between the proposed constitutive formulations of polymer shells in terms the shear modulus parameter. The potential influence of other agents in close proximity on the subharmonic response and threshold is highlighted with respect to the phase interaction between coupled agents.

**2:15**

**1pBA6. *In vitro* acoustic characterization of echogenic liposomes with a polymerized lipid bilayer (Pol-ELIPs).** Shirshendu Paul (Mech. Eng., Univ. of Delaware, 130 Acad.. St., Newark, DE 19716, spaul@udel.edu), Rahul Nahira, Sanku Mallik (Pharmacy, North Dakota State Univ., Fargo, ND), and Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., Washington, Virginia)

Liposomes are typically lipid bilayer vesicles with an aqueous interior. A modified preparation protocol can make them echogenic, i.e., capable of scattering incident acoustic pulses, by incorporating gas pockets in their structure. Here we study echogenic liposomes with a polymerized lipid bilayer that can potentially improve their stability and nonlinear behavior. The Pol-ELIPs prepared were found to have a very polydisperse size distribution with an average size of 3$\mu$m. The frequency dependent attenuation experiment did not show any distinct peak due to the high polydispersity. They showed echogenicity in our *in vitro* scattering experiments, even without the presence of bovine serum albumin in the reconstituting media. Scattered response measured from Pol-ELIP suspension showed non-linear behaviors with distinct second-harmonic and subharmonic peaks, in contrast to non-polymerize ELIPs that did not show any subharmonic response. Scattered fundamental, second-harmonic and subharmonic responses at a lipid concentration of 1µg/ml showed up to 35, 30, and 35 dB enhancement, respectively. The subharmonic response showed all its characteristic features—its appearance only above a threshold excitation level (150 kPa) and then a sharp rise with a subsequent saturation. The study proves the echogenicity of the novel Pol-ELIPs with interesting nonlinear properties. [Work partially supported by NSF.]

**2:30**

**1pBA7. Effects of ambient pressure variation on the subharmonic response from contrast microbubbles: Effects of encapsulation.** Nima Mobaderasny (Mech. and Aerosp. Eng., George Washington Univ., 802 22nd St. NW, Phillips Hall 738, Washington, DC, sany@gwu.edu) and Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., Washington, Virginia)

We investigate the ambient pressure dependent subharmonic response from encapsulated contrast microbubbles for non-invasive estimation of local blood pressure. We have previously found that the subharmonic response from a free microbubble can either increase or decrease or vary nonmonotonically [Katiyar *et al*., J. Acoust. Soc. Am. **129**, 2325–2335] depending on the ratio of the excitation frequency to the resonance frequency (f/f0). Here, we extend this work to encapsulated microbubbles assuming various interfacial rheological models for the encapsulation. With an exponential elasticity model, the general trend remains similar to that of the free bubble. However, one also obtains chaotic oscillations at very low excitation frequencies and smaller damping coefficients. The specific trends are also disrupted when excitation pressure is increased or bubble size is changed. In the talk we will discuss the effects of variations in radius and different parameters of the encapsulation. [Work partially supported by NSF.]

**2:45**

**1pBA8. Probing the bioeffects of cavitation at the single-cell level.** Fang Yuan, Georgy Sankin, Chen Yang, and Pei Zhong (Mech. Eng. & Mater. Sci., Duke Univ., Science Dr., Durham, NC 27713, fang.yuan@duke.edu)

Cavitation induced bioeffects has not been resolved satisfactorily due to the randomness in the inception and bubble dynamics produced by ultrasound. We have developed a microfluidic system to observe consistently the interaction of laser-generated tandem bubbles (50 $\mu$m in diameter) with resultant jet formation, cell deformation, and localized membrane rupture with progressive diffusion of propodium iodide (PI) into individual HeLa cells placed nearby. We observe a clear stand-off distance (SD) dependence in the bioeffects produced by the tandem bubbles. At SD of 10 $\mu$m, all cells underwent necrosis with high, unsaturated level of PI uptake. At SD of 20~30 $\mu$m, 58 to 80% of the cells showed repairable membrane poration with low to medium but saturated level of PI uptake. Within this range, the sub-population of cells that survived without apoptosis increased from ~9% at SD of 20 $\mu$m to ~70% at SD of 30 $\mu$m. The maximum PI uptake, pore size, and estimated membrane strain, however, could vary by more than an order of magnitude at each SD. At SD of 40 $\mu$m, no detectable PI uptake was observed. This experimental system provides a unique tool to probe the bioeffects of cavitation at the single cell level.

**3:00**

**1pBA9. Magnetic targeting of microbubbles at physiologically relevant flow rates.** Joshua W. Owen, Paul Rademeyer, and Eleanor Stride (Dept. of Eng. Sci., Univ. of Oxford, Inst. of Biomedical Eng., Old Road Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, joshua.owen@eng.ox.ac.uk)

The localization of microbubbles to a target site has been shown to be essential to their effectiveness in ultrasound mediated drug delivery and gene therapy. The incorporation of super paramagnetic nanoparticles into the microbubble coating enables them to be manipulated using an externally applied magnetic field. Magnetic microbubbles have been shown to be effective in therapeutic delivery both *in vitro* and *in vivo* in a mouse model. The aim of this experiment was to determine under what conditions in the human body magnetic microbubbles can be successfully imaged and targeted. Different flow rates and shear rates were generated in a tissue mimicking phantom and targeting was observed using a 9.4 MHz ultrasound imaging probe. For the highest shear rates, targeting was also observed optically. Results indicate that magnetic microbubbles can be successfully targeted at shear rates found in the human capillary system (>1000/s) and at flow rates found in the veins and smaller arteries (~200 ml/s). Successful retention was also demonstrated in a perfused porcine liver model simulating conditions *in vivo*. This study provides further evidence for the potential of magnetic microbubbles for targeted therapeutic delivery.

3:15

**1pBA10. Device for detection of cavitation in by spectral analysis in megasonic cleaning.** Claudio I. Zanelli, Samuel M. Howard, Dushyanth Giridhar, and Petrie Yam (Onda Corp., 1290 Hammerwood Dr., Sunnyvale, CA 94089, cz@ondacorp.com)

The authors present a device and a method to detect cavitation in megasonic cleaning environments. The device is small enough to fit between wafers in the semiconductor industry, allowing the monitoring of the cleaning process.

3:30

**1pBA11. A meshless bubble filter for an extracorporeal circulation using acoustic radiation force.** Koji Mino, Manami Kataoka, Kenji Yoshida (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, dmm1017@mail4.doshisha.ac.jp), Daisuke Koyama (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), Kentaro Nakamura (Precision and Intelligence Lab., Tokyo Inst. of Technol., Yokohama, Kanagawa, Japan), Masayoshi Omori, Shigeki Kawarabata, Masafumi Sato (Central Res. Lab. JMS Co., Ltd., Hiroshima, Hiroshima, Japan), and Yoshiaki Watanabe (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan)

Arterial filters are employed in extracorporeal circulations to remove microbubbles and thrombus from the blood flow and prevent the emboli. Filters with mesh structure have a risk to generate the thrombi when the blood flows through it. In this report, a meshless filter using ultrasound is discussed. The filter consists of an aluminum cylinder (length: 130 mm; inner diameter: 30 mm) and two annular ultrasound PZT transducers. The filter has one inlet at the center of the side and two outlets at both ends. By exciting the transducer, the acoustic traveling wave can be generated in the liquid inside the filter. Air bubbles flowing from the inlet can be led toward the outlet by acoustic radiation force. The characteristics of the filter were investigated through a circulation system using distilled water at the driving frequencies of 200 kHz and 1 MHz. Flow and injected air were set as 5.0 l/min and 10 ml/min, respectively. The microbubbles were filtered by using ultrasound and the amount of filtered bubbles was increased with the input voltage to the transducer: 50.6 and 53.7% of microbubbles were filtered at 200 kHz and 1 MHz, respectively, when the input voltage was 100 Vpp.



MONDAY AFTERNOON, 2 DECEMBER 2013 CONTINENTAL 1, 1:00 P.M. TO 4:15 P.M.

# Session 1pMU

## Musical Acoustics: General Musical Acoustics

Randy Worland, Chair

*Physics, Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416*

### *Contributed Papers*

1:00

**1pMU1. Digital fabrication of vocal tract models from magnetic resonance imaging during expert pitch bending on the harmonica.** John Granzow (CCRMA, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, johknee5@gmail.com), Peter Egbert (Ophthalmology, Stanford Univ., Palo Alto, CA), David Barrett (CCRMA, Stanford Univ., San Jose, CA), Thomas Rossing (CCRMA, Stanford Univ., Palo Alto, CA), and Lewis Shin (Radiology, Stanford Univ., Palo Alto, CA)

Expressive pitch bending on the harmonica requires the acoustic coupling of contiguous free reeds, an effect known to arise from highly constrained vocal postures. These techniques have been difficult to demonstrate and instructors often rely on associated vowels to help novices achieve the required constriction of the vocal tract. Magnetic Resonance Imaging (MRI) was recently used to expose the precise vocal contours involved in such expert pitch bends (Egbert, *et al.*, J. Acoust. Soc. Am. **133**, 3590 (2013). To further this investigation, we process the MRI data using 3D slicing software, generating digital models of the vocal tract during sustained bends. In addition to providing volumetric data, the models are also fabricated with fusion deposition modeling and tested on real harmonicas. These tests reveal error tolerances in the conversion from MRI slices to 3d printed models when working with geometries that are highly constrained by a desired acoustic output. Furthermore, comparisons between human performance and simulated output provide clues to the contribution of factors not reproduced in the plastic models such as pharynx dilation.

1:15

**1pMU2. Attack transients in free reed instruments.** Jennifer Biernat (Mansfield Univ. of Pennsylvania, 172 South Broad St., Nazareth, PA 18064, jenbiernat@yahoo.com) and James P. Cottingham (Physics, Coe College, Cedar Rapids, IA)

Attack transients of harmonium-type reeds from American reed organs have been studied in a laboratory setting with the reeds mounted on a wind chamber. Several methods were used to initiate the attack transients of the reeds, and the resulting displacement and velocity waveforms were recorded using a laser vibrometer system and electronic proximity sensors. The most realistic procedure had a pallet valve mechanism simulating the initiation of an attack transient that depressing an organ key would provide. Growth rates in vibrational amplitude were then measured over a range of blowing pressures. Although the fundamental transverse mode is dominant in free reed oscillation, the possibility of higher transverse modes and torsional modes being present in transient oscillation was also explored. The reeds studied are designed with a spoon-shaped curvature and a slight twist at the free end of the reed tongue, intended to provide a more prompt response, especially for larger, lower-pitched reeds for which a slow attack can be a problem. The effectiveness of this design has been explored by comparing these reeds with equivalent reeds without this feature. [Work supported by National Science Foundation REU Grant PHY-1004860.]

1:30

**1pMU3. A study of oboe reed construction.** Julia Gjebic, Karen Gipson (Physics and Music, Grand Valley State Univ., 1 Campus Dr., 118 Padnos Hall, Allendale, MI 49401, gjebicj@mail.gvsu.edu), and Marlen Vavrikova (Music, Grand Valley State Univ., Allendale, MI)

The construction of reeds is of much interest in the oboe community, because professional oboists spend as much time making reeds as they do practicing. Each oboist uses an individual methodology resulting from different training and personal physiology. To investigate how different reed construction affects the resulting sound, 22 professional oboists were recruited to make three reeds apiece for this study. First, a controlled batch of reed cane (internodes of the grass *Arundo Donax*) was selected based on microscopic inspection of cellular composition as well as macroscopic physical attributes. For most of the participants, the cane was then processed identically to the stage known as a *blank*, after which the participants finished their reeds according to their usual methods. (The few participants who made their own blanks still used the controlled cane and also a controlled *staple*, the metal cylinder that attaches the reed to the oboe.) The sound spectra of recordings of each participant playing on his/her respective reeds were analyzed, as was a spectrum of the *crow* (sound without the oboe attached) of each reed in an anechoic chamber. These spectra were correlated to measured physical characteristics of the reeds.

1:45

**1pMU4. Spectral character of the resonator guitar.** Daniel Ludwigsen (Kettering Univ., 1700 University Ave., Flint, MI 49504, dludwigs@kettering.edu)

The resonator guitar was invented in the 1920s, with one or more metal cone resonators set into the body. These additions were originally meant to amplify the sound of the acoustic guitar for performance in a band. The distinct timbre of the resonator ensured that the design survived even after electrification, especially in blues and bluegrass genres. A study of the sound radiated from different models of resonator guitars, as well as a similar standard acoustic guitar, compares spectral features to understand the unique sound of the resonator guitar.

2:00

**1pMU5. A method for obtaining high-resolution directivities from the live performance of musical instruments.** Nicholas J. Eyring (Dept. of Phys. and Astronomy, Brigham Young Univ., 485 N 450 East, Orem, UT 84097, eyringj@gmail.com), Timothy W. Leishman, and William J. Strong (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Directivity measurements for live performance of musical instruments present several experimental challenges, including the need for musicians to play consistently and reproducibly. Some researchers have chosen to implement fixed, limited-element microphone arrays surrounding instruments for rough directivity assessments. Unfortunately, with practical numbers of microphones and data acquisition channels, this approach limits spatial resolution and field decomposition bandwidth. Higher-resolution data may be obtained with a given microphone and channel count by rotating a musician in sequential azimuthal angle increments under a fixed semicircular microphone array. The musician plays a selected note sequence with each increment, but corrections must be made for playing variability. This paper explores the development of this method, which also uses rotating reference frame microphones and frequency response function measurements. The initial developments involve a loudspeaker, with known directivity, to simulate a live musician. It radiates both idealized signals and anechoic recordings of musical instruments with random variations in amplitude. The presentation will discuss how one can reconstruct correct source directivities from such signals and the importance of reference microphone placement when using frequency response functions. It will also introduce the concept of coherence maps as tools to establish directivity confidence.

2:15

**1pMU6. Difference thresholds for melodic pitch intervals.** Carolyn M. McClaskey (Cognit. Sci., Univ. of California, Irvine, 4308 Palo Verde Rd., Irvine, CA 92617-4321, carolyn.mcclaskey@gmail.com)

Pitch-interval processing is an important aspect of both speech and music perception. The current study investigated the extent to which relative pitch processing differs between intervals of the western musical system and whether these differences can be accounted for by the simplicity of an interval's integer-ratio. Pitch-interval discrimination thresholds were measured using adaptive psychophysics for sequentially presented pure-tone intervals with standard distances of 1 semitone (minor second, 16:15), 6 semitones (the tritone, 45:32), and 7 semitones (perfect fifth, 3:2) at both high (1500–5000 Hz) and low (100–500 Hz) frequency regions. Results show similar thresholds across all three interval distances with no significant difference between low and high frequency regions. Consistent with previous studies, thresholds obtained from musicians were considerably lower than those from non-musicians. Data support enhanced pitch-interval perception by musicians but argue against an effect of frequency-ratio simplicity in the case of pure-tone melodic intervals.

2:30

**1pMU7. Analyzing the time variance of orchestral instrument directivities.** Adam T. Buck and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, atbuck@unomaha.edu)

Thirteen-channel anechoic recordings of 20-s musical excerpts for violin, flute, and bass trombone were analyzed to determine how significantly an instrument's directivity changes as it is played. A clearer understanding of the time variance of instrument directivities may help in the advancement of room acoustical computer modeling, as the source properties are crucial to the simulation's accuracy. Previous research has accurately documented the static spatial characteristics of instrument directivities. A multi-channel auralization technique incorporating time-varying source behavior has been developed, but the time variance of the directivities has yet to be explored. In this project, a time-windowing technique was utilized to calculate the directivity index throughout each channel recording for each instrument. The results were analyzed in terms of maximum directivity index and a sampling of complete directivity patterns, and finally used to explore quantification methods. The time variations displayed by each instrument's directivity were unique in terms of magnitude, direction, and frequency. Calculating the average change in directivity index for each channel at each frequency band was found to be a suitable method for summarizing the results. [Work supported by a UNL Undergraduate Creative Activities and Research Experience Grant.]

2:45

**1pMU8. Testing a variety of features for music mood recognition.** Bozena Kostek and Magdalena Plewa (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioacoustics.org)

Music collections are organized in a very different way depending on a target, number of songs or a distribution method, etc. One of the high-level feature, which can be useful and intuitive for listeners, is "mood." Even if it seems to be the easiest way to describe music for people who are non-experts, it is very difficult to find the exact correlation between physical features and perceived impressions. The paper presents experiments aimed at testing a variety of low-level features dedicated to music mood recognition. Musical excerpts to be tested comprise individual (solo) tracks and mixes of these tracks. First FFT- and wavelet-based analyses, performed on musical excerpts, are shown. A set of "energy-based" parameters is then proposed. These are mainly rms coefficients normalized over the total energy derived from wavelet- based decomposed subbands, variance and some statistical moments. They are then incorporated into the feature vector describing music mood. Further part of experiments consists in testing to what extent these features are correlated to the given music mood. Results of the experiments are shown as well as the correlation analysis between two main mood dimensions—Valence and Arousal assigned to music excerpts during the subjective tests.

**3:00–3:15 Break**

**3:15**

**1pMU9. Individuals with congenital amusia respond to fractal music differently from normal listeners.** Fang Liu (Dept. of Linguist. and Modern Lang., The Chinese Univ. of Hong Kong, Rm. G36, Leung Kau Kui Bldg., Shatin N.T., Hong Kong, fangliufangliu@gmail.com), Sherri L. Livengood (The Roxelyn and Richard Pepper Dept. of Communication Sci. & Disord., Northwestern Univ., Evanston, IL), Cunmei Jiang (Music College, Shanghai Normal Univ., Shanghai, China), Alice H. Chan (Div. of Linguist. and Multilingual Studies, School of Humanities and Social Sci., Nanyang Technolog. Univ., Singapore, Singapore), and Patrick C. Wong (Dept. of Linguist. and Modern Lang., The Chinese Univ. of Hong Kong, Hong Kong, China)

Congenital amusia is a neurogenetic disorder predominately defined by impaired perception of musical tonal relationships. This study examined amusics' responses to a gradient of pitch interval complexity in fractal music. Eighteen Mandarin-speaking amusics and 18 controls rated random tone sequences for perceptual (complexity, melodicity) and affective (interest, ease, mood) attributes, and performed a recognition memory task. Sequences were created using fractal model ($1/f^{ß}$) with ß-values ranging from 0.0 (most complex) to 2.6 (least complex). As predicted, both groups rated complexity based on the ß-values, demonstrating that amusics perceived the gradient of pitch interval complexity. However, amusics' ratings deviated from controls in measures of melodicity, affect, and memory performance. For controls, moderately complex sequences (fractal ß-values = 1.4–1.6) were rated the most melodious, drove the highest emotional responses, and were the easiest to remember, whereas amusics' ratings did not respond to this range, but rather followed a more linear trend. These findings suggest that amusics not only have problems with perception of pitch interval relationships (not complexity), but also lack heightened sensitivity to the moderate range in fractal music. This deficit is reflected in broader musical processing including the perception of melody, affective response, and memory for musical sequences.
[This work was supported by National Science Foundation Grant BCS-0719666 to P.C.M.W. and Shanghai Normal University funding to C.J.]

**3:30**

**1pMU10. Marching band hearing responses: Indoor rehearsal vs outdoor performance configurations.** Glenn E. Sweitzer (Sweitzer LLP, 4504 N Hereford Dr., Muncie, IN 47304, glenn.sweitzer@gmail.com)

Marching band performer responses are gathered to compare how well each performer hears: (1) oneself playing; (2) others playing the same part and instrument; and (3) others playing different parts, by instrument. The measurement protocol is repeated in an indoor purpose-built band rehearsal venue with adjustable sound diffusive vs absorptive wall treatments and, at an open-air outdoor performance venue (grass playing field). All scaled responses are gathered using a personal response system, providing immediate, simultaneous, and anonymous responses that can be compared online. Prior to each set of responses, all performers play together a well-rehearsed score. Responses vary indoors by absorptive treatment and band configuration and, outdoors, by band configuration. Discussion will focus on how marching bands, indoors or outdoors, might be reconfigured to improve hearing for performers, directors, and audiences.

**3:45**

**1pMU11. Representation of musical performance "grammar" using probabilistic graphical models.** Gang Ren, Zhe Wen, Xuchen Yang, Cheng Shu, Fangyu Ke, and Mark Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu)

As a versatile data modeling tool, probabilistic graphical model can be applied to model the complex dependency structures encoded in the contextual "grammar" of music performance. The musical performance grammar here refers to the relational structures of the sonic features extracted from music performances. In the existing literature, the data structure of musical expressive grammar is usually modeled as rule list, following the grammatical format of natural language processing applications. In this work, we apply the representation format of probabilistic graphical model to musical performance features to extend the conventional rule-list format. We choose probabilistic graphical model as an "upgraded" representation for two reasons. First, probabilistic graphical model provides enhanced representation capability of relational structures. This feature enables us to model the complex dependency structure that the conventionally rule list cannot handle. Second, the graphical format of probabilistic graphical model provides an intuitive human-data interface and allows in-depth data visualization, analysis, and interaction. We include the representation and analysis examples of musical performance grammar obtained from both manual analysis and automatic induction. We also implemented interpretation tools that interface the rule-list format and the probabilistic graphical model format to enable detailed comparison with existing results of musical performance analysis.

**4:00**

**1pMU12. Vibrato lab: A signal processing toolbox for woodwind sound analysis.** Zhe Wen, Xuchen Yang, Cheng Shu, Fangyu Ke, Gang Ren, and Mark Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, g.ren@rochester.edu)

Vibrato is an important performance technique for woodwind instrument that produces amplitude and frequency modulation inside a musical note. We present a MATLAB-based toolbox for detailed analysis of recorded vibrato notes. The harmonic structure of a music note is identified from its spectrographic analysis. Then we separate the harmonic structure into sonic partials using band-passed filters. The sound analysis algorithms, which provide the signal features of the vibrato note, are then performed over these separated sonic partials. In our implementation, each individual sonic partial is modeled as a quasi-monochromatic component, which is a sinusoidal signal with narrow-band amplitude modulation and frequency modulation. Based on this signal modeling technique, the modulating components are extracted from a separated sonic partial using modulation detection algorithms including short-time analysis and Hilbert transform. This toolbox provides comprehensive visualization tools to allow the users to interact and experiment with these signal features. We also implemented an auralization module that allows the users to experiment with the signal parameters and synthesize artificial vibrato sound. The analysis and visualization functionalities are all integrated in a compact graphical user interface that allows the users to intuitively implement complex sound analysis functionalities with simple analysis procedures.

## Session 1pPA

# Physical Acoustics, Animal Bioacoustics, and Signal Processing in Acoustics: Nonlinear Sound and Ultrasound Field Reconstruction and Related Applications II

Yong-Joe Kim, Cochair
*Texas A&M Univ., 3123 TAMU, College Station, TX 77843*

Je-Heon Han, Cochair
*Mech. Eng., Texas A&M Univ., 3123 TAMU, College Station, TX 77840*

### *Invited Papers*

**1:00**

**1pPA1. Nonlinear and transient acoustic holography for characterization of medical ultrasound sources and their fields.** Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow St. Univ., and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Leninskie Gory, Moscow 119991, Russian Federation, oleg@acs366.phys.msu.ru), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), and Sergey A. Tsysar (Dept. of Acoust., Phys. Faculty, Moscow St. Univ., Moscow, Russian Federation)

Holography is based on the possibility of reproducing a 3D wave field from a 2D distribution of the wave amplitude and phase measured along some surface transverse to the wave propagation. Such a measured distribution thus can be considered as a hologram. It provides a boundary condition for the wave equation, and as such is an important characteristic of any ultrasound source. In our previous work we have implemented various holographic approaches for medical ultrasound sources, including transient and nonlinear versions. Here we illustrate the approach with several experimental examples. Transient holography was performed to characterize the surface vibration of a circular, single-element, flat diagnostic probe with a 2.5 cm diameter and a 1 MHz resonance frequency after excitation by a submicrosecond pulse. Nonlinear holography was applied to a single-element focused source with a diameter and focal length of 100 mm. Forward- and backward-propagation algorithms were based on the Westervelt wave equation. A single-element focused source was excited at a sufficiently high power level so that harmonics were developed during nonlinear propagation in water; the nonlinear hologram was recorded as a set of 2D distributions of magnitude and phase for several harmonics. [Work supported by RFBR and NIH EB007643.]

**1:20**

**1pPA2. Crack detection in long rod by impact wave modulation method.** Alexander Sutin (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu), Richard Haskins, and James Evans (Engineer Res. and Development Ctr., U.S. Army Corp of Engineers, Vicksburg, MS)

Engineering Research and Development Center (ERDC) in Vicksburg, MS, is interested in methods of microcracks detection in extended length and conducted test of applications of nonlinear wave modulation spectroscopy (NWMS). This is one of the simplest methods of nonlinear acoustic NDE. It is based on measurements of the modulation of a high frequency wave by a low frequency vibration. NWMS can detect the crack presence but cannot localize cracks. We present modification of NMWS method based on the modulation of by ultrasound by short pulse produced by impact. This method allows crack and damage localization using time delay of the impact produced pulse and modulated part of high frequency wave. The feasibility test was conducted for 60 ft long steel trunnion rod with an imitated crack. The continuous wave ultrasound with the frequency about 50 kHz was modulated by the longitudinal impulse produced by the hammer impact. Time delay between the impact and the received modulated ultrasonic wave allowed finding of the distance to the crack. Propagation speed of the modulated wave was lower than the speed of the hammer impulse as it is followed from the theory describing frequency dispersion of the waves in rods. [The project is funded by the Navigation Research Program at ERDC.]

**1:40**

**1pPA3. Transient nonlinear acoustical holography.** Yun Jing (Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695, yjing2@ncsu.edu) and Jonathan Cannata (HistoSonics Inc., Ann Arbor, MI)

This paper presents our recent work on transient nonlinear acoustical holography. A higher order stepping algorithm is first introduced, which is shown to be significantly more accurate and efficient than the original one (Evaluation of a wave-vector-frequency-domain method for nonlinear wave propagation, Jing *et al*., J. Acoust. Soc. Am. **129**, 32) through systematic numerical study. Underwater experimental results from a highly focused transducer will be presented here to show the validity of the model. Both linear and nonlinear, forward and backward projection of the acoustic field are conducted. While linear acoustical holography is shown to produce erroneous results, good agreement is found between our nonlinear model and the experiment.

2:00

**1pPA4. Possibilities of tomography system prototype using third-order acoustic nonlinear effects.** Valentin A. Burov, Roman V. Kryukov, Andrey A. Shmelev, Dmitry I. Zotov, and Olga D. Rumyantseva (Dept. of Acoust., Moscow State Univ., Faculty of Phys., Leninskie Gory, GSP-1, Moscow, Russia, Moscow 119991, Russian Federation, blackrainbow13@mail.ru)

Third-order nonlinear acoustical tomography is very important for medical diagnostics, because it will provide information on the new and unexplored quantity—third-order nonlinear acoustical parameter. A prototype of the tomography system for reconstructing the distributions of the acoustic nonlinear parameters is developed in our group on the basis of effect of nonlinear noncollinear interaction of three primary waves. Application of coded primary signals with further correlation processing of a detected signal at combination frequencies makes it possible to reconstruct the complete image of an object using three transmitters and one receiver. Third-order effects are born from two competing processes—the pure third order interaction (informational for diagnostic purposes) and the twofold interaction of the second order (interfering). To explore the application boundaries of the third-order nonlinear acoustical tomography, these two competing processes are considered in details. Two cases are compared. In the first case, the interaction takes place between two broadband coded signals and one monochromatic signal. In the second case, all three primary signals are broadband coded; then, amplitude of the interfering part of the signal is smaller. Moreover, in this case, the possibility of reconstructing the spatial distribution of the second and third-order nonlinear acoustical parameters arises due to the reciprocity principle generalized on the nonlinear scattering processes.



### *Contributed Papers*

2:20

**1pPA5. Second-harmonic generation in shear wave beams with different polarizations.** Kyle S. Spratt, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, 4307 Caswell Ave. Apt. E, Austin, TX 78751, sprattkyle@gmail.com)

A parabolic equation describing the propagation of collimated shear wave beams in isotropic elastic solids was derived by Zabolotskaya [Sov. Phys. Acoust. **32**, 296–299 (1986)], and was seen to contain both cubic and quadratic nonlinear terms at leading order. While second-order nonlinear effects vanish for the quasi-planar case of linearly-polarized shear wave beams, the importance of quadratic nonlinearity for more complicated polarizations is not yet well understood. The current work investigates the significance of quadratic nonlinearity by considering second-harmonic generation in shear wave beams generated by a certain class of source polarizations that includes such cases as radial and torsional polarization, among others. Corresponding to such beams with Gaussian amplitude shading, analytic solutions are derived for the propagated beam at the source frequency and the second harmonic. Diffraction characteristics are discussed, and special attention is paid to the relationship between the source polarization of the beam and the polarization of the subsequently generated second harmonic. Finally, suggestions are made for possible experiments that could be performed in tissue phantoms, exploiting the theoretical results of this work. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

2:35

**1pPA6. Average radiation force at high intensity: Measured data.** Nick V. Solokhin (Ultrasonic S-Lab, 3628 Clayton Rd. # 102, Concord, CA 94521, solokhin@comcast.net)

Measurements were done in water at room conditions with HIFU transducer (at frequencies 3.1 and 4.3 MHz, max. pressure amplitude was 6—8 MPa). Average radiation force (ARF) was measured with flat reflecting target at normal incidence. The target was moved along the acoustic axis of the transducer: distance varied from 0.7 F to 1.4 F (F is focal distance). Measured ARF was growing with the distance (~25%) and it got max at distance 1.2 F. This effect meets with growing of nonlinear distortions at growing of intensity and length of passed way. It was measured dependence of ARF upon the angle of incidence and with same target. Measurements were done at incident angles 0, 30, and 60 degree. ARF changed from max value (at 0 angle) and reduced to 0.5 max value at 60 degree.

MONDAY AFTERNOON, 2 DECEMBER 2013 CONTINENTAL 9, 1:30 P.M. TO 3:00 P.M.

## Session 1pSPa

## Signal Processing in Acoustics: Acoustical Imaging and Sound Field Reconstruction

David Chambers, Chair
*Lawrence Livermore National Lab., PO Box 808, L-154, Livermore, CA 94551*

### *Contributed Papers*

1:30

**1pSPa1. Dynamic acoustical imaging systems with reconfigurable transceiver arrays and probing waveforms.** Michael Lee and Hua Lee (Elec. and Comput. Eng., Univ. of California, 3121 Frank Hall, Santa Barbara, CA 93106, hualee@ece.ucsb.edu)

The data-acquisition format of traditional acoustical imaging systems has been operating with structured transceiver arrays and predetermined illumination waveforms. The fixed physical array configurations have been largely linear or planar, for computation simplicity. The conventional systems started with the coherent mode with narrow-band illumination. The coherent illumination waveforms were then replaced by wideband signals to operate in the pulse-echo format for the improvement of range resolution. This paper presents the functionalities of reconfigurable acoustical imaging systems with FMCW probing waveforms, and the equivalence to the conventional modalities. The reconfigurable configuration is applied to both the transceiver aperture arrays and the probing signals for the enhancement of the resolving capability. The reconfigurable array allows us to actively

optimize the aperture coverage for superior resolution. To achieve reconfigurable illumination waveforms, programmable stepped-frequency FMCW signaling modality is employed. The presentation of this paper includes the algorithm structure, range estimation, and frequency editing, and of special interest and importance, the detection and imaging of time-varying targets.

**1:45**

**1pSPa2. Reconstruction of arbitrary sound fields with a rigid-sphere microphone array.** Efren Fernandez-Grande (Acoust. Technol., DTU, Tech. Univ. of Denmark, Ørsteds Plads, B. 352, DTU, Kgs. Lyngby DK-2800, Denmark, efg@elektro.dtu.dk)

Over the last few years, several studies have examined the potential of using rigid-sphere microphone arrays for reconstructing sound fields with near-field acoustic holography (NAH). The existing methods provide a reconstruction of the sound field based on a spherical harmonic expansion. However, because of the basis functions used, the reconstruction can only be performed in spherical surfaces concentric to the array and inside the source-free region, which imposes a severe limitation on the applicability of the technique due to geometrical constrains. In this paper, a method based on an equivalent source model is proposed, where a combination of point sources is used to describe the incident sound field on the array. This method makes it possible to reconstruct the entire sound field at any point of the source-free domain without being restricted to a spherical surface. Additionally, this approach adds versatility (the reconstruction can be based on the microphones that are closer to the source, for a better conditioning of the problem, or also use non-uniform sampling). The method is presented, examined numerically and experimentally, and compared to the existing methods based on a spherical harmonic expansion.

**2:00**

**1pSPa3. Three-dimensional reconstruction of sound fields based on the acousto-optic effect.** Efren Fernandez-Grande (Acoust. Technol., DTU, Tech. Univ. of Denmark, Ørsteds Plads, B. 352, DTU, Kgs. Lyngby DK-2800, Denmark, efg@elektro.dtu.dk) and Antoni Torras-Rosell (DFM, Danish National Metrol.Inst., Kgs. Lyngby, Denmark)

The acousto-optic effect can be used to measure the pressure fluctuations in air created by acoustic disturbances (the propagation of light is affected by changes in the medium due to the presence of sound waves). This makes it possible to measure an arbitrary sound field using acousto-optic tomography via scanning the field with a laser Doppler vibrometer. Consequently, the spatial characteristics of the sound field are captured in the measurement, implicitly bearing the potential for a full holographic reconstruction in a three-dimensional space. Recent studies have examined the reconstruction of sound pressure fields from acousto-optic measurements in the audible frequency range, based on Fourier transforms and elementary wave expansion methods. The present study examines the complete reconstruction of the sound field from acousto-optic measurements, recovering all acoustic quantities, and compares the results to the ones obtained from conventional microphone array measurements.

**2:15**

**1pSPa4. Put your sound where it belongs: Numerical optimization of sound systems.** Stefan Feistel (Ahnert Feistel Media Group, Berlin, Germany), Bruce C. Olson (Ahnert Feistel Media Group, Brooklyn Park, Minnesota), and Ana M. Jaramillo (Ahnert Feistel Media Group, 3711 Lake Dr., Robbinsdale, Minnesota 55422, anaja@vt.edu)

A new technology based on FIR filters in combination with room acoustic modeling allows optimizing steerable columns and line arrays to each specific venue in a matter of seconds. Both maximum SPL and maximum sound field uniformity can be prioritized to obtain an ideal sound distribution without losing sound pressure in the wrong directions thus avoiding unwanted reflections. Areas can also be excluded to minimize sound (i.e., stage area). Real-life test results support the theory.

**2:30**

**1pSPa5. Multipole, spherical harmonics and integral equation for sound field reproduction.** Jung-Woo Choi (Mech. Eng., KAIST, YuseongGu GuseongDong 373-1, Daejeon 373-1, South Korea, khepera@kaist.ac.kr)

The quality of sound field reproduction depends on the way we use to represent a sound field. For example, the spherical harmonics expansion being used for the higher-order-Ambisonics attempts to represent a desired sound field as a sum of many spherical harmonics, and the method incorporating integral equations, e.g., wave field synthesis, identifies the sound field as a superposition of single or double layer potentials distributed on a boundary surface. In contrast, the multipole expansion converts the desired sound field into equivalent multipole distributions. In this work, we investigate the fundamental differences in these three representations when they are applied for the reproduction of sound fields. In particular, their advantages and disadvantages in representing the directivity of a virtual sound source, translation of a desired sound field, and their benefit in deriving time domain formula for the real-time application will be discussed.

**2:45**

**1pSPa6. The use of interpolated time-domain equivalent source method for reconstruction of semi-free transient sound field.** Siwei Pan and Weikang Jiang (State Key Lab. of Mech. Syst. and Vib., Shanghai Jiao Tong Univ., 800 Dongchuan Rd., Shanghai 200240, China, swpan@sjtu.edu.cn)

A semi-free transient sound field is reconstructed by extending the interpolated time-domain equivalent source method (ITDESM) in the free field to the semi-free field. In this approach, the time-domain equivalent sources are placed not only near the actual sound sources, but also in the vicinity of their mirrored sources with respect to the reflecting plane surface. Suppose that the number of equivalent sources distributed around the mirrored sources (virtual equivalent sources) is the same as that used near the actual sources (actual equivalent sources), with their locations symmetrical about the reflecting surface. The reflecting surface considered here can be perfectly rigid as well as impedance-effected. Furthermore, by reformulating the strengths of the virtual equivalent sources at each time instant with those of the corresponding actual equivalent sources, the computation load of solving the equivalent source strengths can be reduced by 50%. Numerical examples of reconstructing the semi-free transient sound field radiated from three monopoles under different reflection conditions demonstrate the feasibility of the proposed method.

## Session 1pSPb

# Signal Processing in Acoustics: Filtering, Estimation, and Modeling in Acoustical Signal Processing

Edmund J. Sullivan, Chair
*Prometheus Inc., 46 Lawton Rook Lane, Portsmouth, RI 02871*



### *Contributed Papers*

**3:15**

**1pSPb1. A new look at the matched filter.** Edmund J. Sullivan and James G. Kelly (Prometheus Inc., 46 Lawton Rook Lane, Portsmouth, RI 02871, ed@prometheus-us.com)

The matched filter is well known as the optimal linear detector. It is generally used in threshold tests, which depend only on its maximum value. However, it is usually used as an estimator, an example being the case of range determination in the active sonar problem. But even in this case, only its peak value is used. Here, we look at what we refer to as the "complete" matched filter. By examining the full output of the matched filter we show that it is closely related to the deconvolution problem, which is not a detector but an estimator. Further, we show that for the case of a linear chirp (LFM) signal, the full matched filter and the deconvolution problem are essentially identical, the difference being in the power spectrum of the signal. The more the signal power spectrum deviates from whiteness, the greater the difference between the two, even if the time-bandwidth product of the signal remains the same. Moreover, we show that the LFM signal is an optimal signal for the deconvolution problem since it minimizes the variance of the estimate.

**3:30**

**1pSPb2. The spectral profile estimation algorithm: a non-linear, non-*a priori* noise normalization algorithm.** Jeffrey A. Ballard (Appl. Res. Labs., The Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, ballard@arlut.utexas.edu)

Time-frequency analysis of acoustic signals often involves a background noise spectral estimation step to estimate the signal-to-noise ratio of the signals of interest. This step is typically accomplished with the application of a split mean normalizer (SPN) [Struzinski and Lowe, J. Acoust. Soc. Am. **76** 1738–1742 (1984)]. These normalizers work well against tones and can be tuned with a priori information to normalize wider energy signals but can still suffer performance loss in these cases. For signal dense spectra that contain both tonal and broadband-like energy the SPN has difficulty separating noise from signal, and the signal energy unduly influences the background estimate. This work introduces a new normalization scheme called the Spectral Profile Estimation (SPE) algorithm, which operates with no a priori information to estimate the background noise of these signal dense spectra. The SPE algorithm assumes that the background noise has an $f^{-m}$, $m>0$, shape, and that signals are superimposed on the noise and are associated with spectra maxima. The SPE algorithm then finds and connects local minima to estimate the background. The SPE algorithm is first explained and then applied to experimental data. Finally, SPE performance is compared to the performance of SPN.

**3:45**

**1pSPb3. Artificial reverberation using multi-portacoustic elements.** Warren L. Koontz (Rochester Inst. of Technol., 159 Coco Palm Dr., Venice, Florida 34292, profwub@gmail.com)

Multi-port acoustic elements, including simple two-port elements, can be used both to model acoustic systems and to create acoustic signal processing structures. This paper focuses on the latter, specifically on networks of multi-port acoustic elements that create an artificial reverberation effect. We will introduce some basic two-port and multi-port building blocks, each characterized by a scattering matrix and demonstrate frequency domain and time domain analyses of networks of these elements. We will then propose and investigate networks that create artificial reverberation. We will implement these structures using MATLAB and evaluate and compare them with some existing approaches including the Schroeder reverberator and feedback delay networks. Comparisons will be based on the computed impulse response as well as sound files.

**4:00**

**1pSPb4. An investigation of the subjective quality of non-linear loudspeaker models employing Volterra series approximations.** Ian Richter (Peabody Inst. of the Johns Hopkins Univ., 606 St. Paul St, Box #14, Baltimore, MD 21202, ian.richter@gmail.com)

This investigation is an extension of the work of Angelo Farina on the use of Volterra-series approximations to model acoustic systems. Farina's method uses a logarithmically swept sine chirp to extract the transfer function of a system, including its harmonic nonlinearities. I have used this approach to construct computer models of several loudspeakers. I then evaluated my models using a blind listening test to compare their outputs against recordings of the actual loudspeakers. The results of the blind test suggest that, while this algorithm is not sufficient to convincingly model the loudspeaker transfer function, it might be useful as part of a multi-stage algorithm.

**4:15**

**1pSPb5. Online sound restoration system for digital library applications.** Andrzej Czyzewski, Janusz Cichowski, Adam Kupryjanow, and Bozena Kostek (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl)

Audio signal processing algorithms were introduced to the new online non-commercial service for audio restoration intended to enhance the content of digitized audio repositories. Missing or distorted audio samples are predicted using neural networks and a specific implementation of the Janssen interpolation method based on the autoregressive model (AR) combined with the iterative restoring of missing signal samples. Since the distortion prediction and compensations algorithms are computationally complex, an implementation which uses parallel computing has been proposed. Many archival recordings are at the same time clipped and affected by wideband noise. To restore those recordings, the algorithm based on the concatenation of signal clipping reduction and spectral expansion was proposed. The clipping reduction algorithm uses an intelligent interpolation to replace distorted samples with the predicted ones based on learning algorithms. Next, spectral expansion is performed in order to reduce the overall level of noise. The online service has been extended with some copyright protection mechanisms. Immunity of watermarks to the sound restoration is discussed with regards to low-level music feature vectors embedded as watermarks. Then, algorithmic issues pertaining watermarking techniques are briefly recalled. The architecture of the designed system together with the employed workflow for embedding and extracting the watermark are described. The implementation phase is presented and the experimental results are reported.

# Session 1pUW

## Underwater Acoustics and Acoustical Oceanography: Deep Water Acoustics II

John Colosi, Cochair
*Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943*

Karim G. Sabra, Cochair
*Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405*

Kathleen E. Wage, Cochair
*George Mason Univ., 4400 University Dr., Fairfax, VA 22030*

### *Invited Papers*

**2:00**

**1pUW1. A comparison of measured and predicted broadband acoustic arrivals in Fram Strait.** Hanne Sagen (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu), Stein Sandven (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Agnieszka Beszczynska-Moeller (Inst. of Oceanol. PAS, Sopot, Poland), Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Florian Geyer (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway), Brian D. Dushaw (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Mohamed Babiker (Nansen Environ. and Remote Sensing Ctr., Bergen, Norway)

Fram Strait is the only deep-water connection between the Arctic and the world oceans. On the eastern side, the northbound West Spitsbergen Current transports warm Atlantic water into the Arctic, while on the western side the southbound East Greenland Current transports sea ice and polar water from the Arctic to the Nordic Seas and Atlantic Ocean. Significant recirculation and intense small-scale mesoscale variability in the center of the Strait make it difficult to accurately measure ocean transports through the Strait. An acoustic system for tomography, glider navigation, and passive listening was installed in the central, deep-water part of the Strait during 2010–2012. The integral measurements of temperature provided by tomography and the spatial resolution of the glider data are complementary to the data from the long-term array of oceanographic moorings at 780 50’N. The oceanographic conditions and highly variable sea ice in Fram Strait provide an acoustic environment that differs from both the high Arctic and the temperate oceans and that results in complex acoustic propagation. Improved understanding of the measured acoustic arrivals through comparison with predictions based on available environmental data is important for development of tomographic inversion and assimilation techniques, for glider navigation, and for acoustic communications.

**2:20**

**1pUW2. Observations at Ascension Island of T-phases from earthquakes in the Fiji-Tonga region.** Mark K. Prior, Mario Zampolli (PTS, Comprehensive Nuclear-Test-Ban Treaty Organisation, PO Box 1200, Vienna 1400, Austria, mark.prior@ctbto.org), and Kevin Heaney (OASIS, Lexington, MA)

Ascension Island in the Atlantic Ocean is the site of one of the hydrophone stations that make up part of the hydroacoustic network operated by the Comprehensive Nuclear-Test-Ban Treaty Organization. Hydrophones are deployed in two groups of three, known as triads; one to the north of the island and one to the south. Correlation processing across hydrophones allows signal azimuths to be determined and both triads show large numbers of signals arriving from the south-west. These signals are generated by earthquakes in the region between Fiji and Tonga in the Pacific Ocean and travel through Drake Passage between Antarctica and South America. Signal azimuths are studied and it is shown that some signals originate from earthquakes to which two-dimensional propagation modeling would suggest there is no direct path. The mechanisms by which sound from these “blocked” regions might reach Ascension Island are discussed.

**2:40**

**1pUW3. Observation and modeling of three-dimensional basin scale acoustics.** Kevin D. Heaney, Richard L. Campbell (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), Mark Prior, and Mario Zampolli (CTBTO, Vienna, Austria)

In this paper, an overview of some of the basin-scale recordings of the International Monitoring System of the Comprehensive Test Ban Treaty will be presenting, including observations of distant earthquakes, under-sea volcanoes and cracking ice-sheets. Long range experiments were conducted in 1960 when nearly antipodal receptions were made at Bermuda from SuS charges deployed off the coast of Australia. Three-dimensional propagation effects was an important part of the propagation. For propagation ranges of thousands of

kilometers, interaction with islands, ridges and seamounts is expected to influence propagation. Previous modeling approaches have been either a fully 3D ray approach, a hybrid adiabatic mode-ray approach and a hybrid adiabatic mode—PE approach. In this paper, we present the a global scale three-dimensional split-step Pade Parabolic Equation. Model results are compared with several direct observations of arrivals that are clearly in the 2D propagation shadow. The impact of 3D propagation on the coverage maps of the CTBTO are significant, relative to current 2D predictions.

## *Contributed Papers*

1p MON. PM

**3:00**

**1pUW4. New formulae for horizontal coherence from path integral theory.** John Colosi (Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

Previously published path integral results for the horizontal coherence length utilized an empirical relation for the phase structure function density that scaled as lag to the three-halves power. Here, a Taylor series expansion is carried out such that the phase structure function density scales instead as the second power of lag, consistent with other path integral coherence scales such as depth and time. The resulting integral equations are solved analytically. The new result shows the expected one over square-root range and one over frequency scalings, and it demonstrates more clearly how transverse coherence is sensitive to the space-time scales of ocean sound-speed perturbations.

**3:15**

**1pUW5. An alternative method for the estimation of underwater acoustic signal coherence.** Matthew Dzieciuch and Peter Worcester (SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225, mad@ucsd.edu)

The estimator-correlator is the optimal method for measuring the travel-time of a scattered underwater tomographic acoustic signal. The method increases the signal-to-noise ratio at some cost in resolution when the signal and noise covariances are known. The noise covariance is easily measured when the signal is not present. The signal covariance may not be known but can be estimated as well. The procedure is to parameterize the signal covariance and find the maximum output signal-to-noise ratio of the estimator-correlator while varying the signal covariance parameters. Following Flatte *et al.*, the path integral model of signal covariance is $R = \exp\{ -(\Delta t/T_c)^2 + (\Delta f/B_c)^2 + (\Delta z/D_c)^2\}$. Thus, a simple search over coherence time, $T_c$, coherence bandwidth, $B_c$, and coherence depth, $D_c$, produces estimates of those parameters. Application of this technique with data from three recent tomographic experiments will demonstrate its efficacy. Time coherence was measured during the 2009 Philippine Sea Experiment at 250 Hz and 190 km range. Vertical coherence was measured during the 2010 Philippine Sea experiment at 250 Hz and at ranges from 120 to 450 km. Horizontal coherence was measured in the North Pacific at 75 Hz and 3500 km range.

**3:30**

**1pUW6. The mode view of long-range propagation through oceanic internal waves.** Frank S. Henyey (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, frank@apl.uw.edu)

For long-range acoustic propagation in the ocean, calculation of internal wave effects by ray tracing is far from accurate. Full wave methods must be used. An unperturbed normal mode simulation has been developed for this case. Simulations have been carried out with a single-mode starting field, with a sound speed profile measured in the 2010 Philippine Sea experiment, and internal waves consistent with measurements at the same region and same time. Modes were chosen that turn around half way between the sound axis and the surface. The frequency is 100 Hz, appropriate for long-range propagation experiments. Ensemble averaged mode intensities are consistent with transport theory results, but the individual mode intensities as a function of range are very inconsistent with a random walk ("diffusion") in mode number space.

**3:45**

**1pUW7. Deep fades in intensity: Exploration of measurement-Monte Carlo parabolic equation mismatch in the Philippine Sea.** Andrew W. White (Earth and Space Sci., Univ. of Washington, 433 31st Ave. E, Seattle, WA 98112, andrew8@apl.washington.edu), Rex K. Andrew, James A. Mercer (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA), John A. Colosi (Oceanography, Naval Postgrad. School, Monterey, CA), Lora J. Van Uffelen, and Bruce M. Howe (School of Ocean and Earth Sci. and Technol., Univ. of Hawaii at Manoa, Honolulu, HI)

The oceanography of the Philippine Sea is partially characterized by energetic mesoscale and strong locally generated internal tides. Despite the simplification of range-independence and the exclusion of internal tides from Monte Carlo parabolic equation (MCPE) simulations, predictions of scintillation index, variance of log-intensity, and the distribution of intensity for acoustic paths with upper-turning-points (UTP) below the extreme upper ocean generally agree with measurements made during an experiment in 2009. These measures of the fluctuations did not appear to be strongly influenced by the number of UTPs in the path, though a compensating effect due to differences in UTP position cannot be ruled out. Enhanced variability in the form of deep fades is observed for paths turning in the extreme upper ocean; this enhanced variability is not predicted by the MCPE model employed. Seaglider-based observations of mixed-layer depth (from 2010 to 2011) and moored measurements of internal-tide-related sound-speed perturbations are presented. A plane-wave internal-tide model and results from acoustic mode propagation through range-independent profiles measured *in situ* are compared with the observed character of the intensity fades.

**Payment of separate registration fee required. See page XXX**

MONDAY AFTERNOON, 2 DECEMBER 2013 CONTINENTAL 7/8, 7:00 P.M. TO 9:00 P.M.

## Session 1eID

## Interdisciplinary: Tutorial Lecture on Time Frequency Analysis: Theory and Applications

R. Lee Culver, Chair
*Pennsylvania State University, State College, PA 16804*

**Chair's Introduction—7:00**

### *Invited Paper*

**7:05**

**1eID1. Time-frequency analysis: Theory and applications.** Leon Cohen (Dept. of Phys., City Univ. of New York, 695 Park Ave., New York, NY 10065-5024, leon.cohen@hunter.cuny.edu) and Patrick Loughlin (Dept. of Bioeng., Univ. of Pittsburgh, Pittsburgh, PA)

Time-varying spectra are one of the most primitive sensations we experience, since we are surrounded by light of changing color, by sounds of varying pitch, and by many other phenomena whose periodicities change. The development of the physical and mathematical ideas needed to explain and understand time-varying spectra has evolved into the field now called "time-frequency analysis." Among the many signals whose frequency content has been shown to vary in time are speech and other animal sounds, biomedical signals (e.g., heart sounds, heart rate, the electroencephalogram (EEG), the electromyogram (EMG), and others), music, radar and sonar signals, and machine vibrations, among others. In this tutorial, we give an overview of time-frequency analysis, with a focus on its applications. We describe how these methods impinge on and clarify issues in biomedical and biological signal analysis, wave propagation, random systems, non-linear systems, and other areas. Of particular interest is the application of time-frequency analysis to pulse propagation in dispersive media. We show that time-frequency considerations lead to new approximation methods for dynamic systems and wave propagation. We describe how to transform wave equations into phase-space, where the resulting equation is often more revealing than the original wave equation. We also discuss the applications to random systems and in particular to the propagation of noise fields.

## Session 2aAA

## Architectural Acoustics and Physical Acoustics: Acoustic Parameters of Materials: Their Definition, Measurement, and Uses in Architectural Acoustics

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

**Chair's Introduction—8:00**



### *Invited Papers*

**8:05**

**2aAA1. A new way of determining the total absorption of gypsumboard wall structures.** Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

Gypsumboard wall construction is used throughout the United States and other parts of the world. Measuring the absorption of wall structures has been done in ways that always leave questions as to the efficacy of the measurement. Walls have been laid down on the floor and measured, and stood straight up and measured. Both methods leave questions as to the data because of differing results. There are questions about diaphragmatic movement that affects the absorption and its constrainment. We are proposing a different method that we think solves the problems inherent in each of the other methods, and we are presenting the results of measurements of indicated materials and comparisons with other methods

**8:25**

**2aAA2. Acoustic diffusers, the effects of materials and finishes on diffusion efficiency and absorption coefficients.** Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

Over the years, many different theories about the effects of material mass, finishes, porosity, and other physical attributes of diffuser design have been put forth. These theories have included excess absorption and/or reduced diffusion being caused by different woods or other materials being used as well as different types of paints and finishes. This paper is a study on the veracity of those theories and the effects of assorted materials in the construction of acoustic diffusers. The study will include the use of a specific diffuser design utilizing a periodic design with specific frequency ranges as well as more simplified geometric designs such as pyramids and barrels. Tests will be conducted to look at specific absorption results and the correlation to diffusion efficiency, if any. Standard testing procedures will be used to derive the information presented in the paper.

**8:45**

**2aAA3. Determining sound transmission through damped partitions: Challenges in theoretical prediction and laboratory testing.** Benjamin Shafer (Serious Energy, Inc., 1117 Tacoma Ave. South, Tacoma, Washington 98402, bshafer@craworld.com)

The sound transmission loss through traditional wall and ceiling building partitions can currently be predicted using software programs and/or laboratory test data. There is, however, a great divide that separates the theoretical prediction of sound transmission loss from the laboratory-measured values. Some of the most common prediction software packages do not account for dynamic (frequency- and temperature-dependent) material properties and are, therefore, incapable of predicting sound transmission accurately for some common solutions. The most accurate and precise software prediction tools available become impractical because the cross-correlation between laboratory testing facilities is so poor that it is not possible to replicate predicted performance dynamics in a laboratory setting. Using the structural properties of damping materials as an example, the divide between theoretical prediction and laboratory testing will be illustrated and possible solutions for closing such a divide will be presented.

**9:05**

**2aAA4. An innovative acoustic muffler to reduce acoustic leakage from recessed lights, intake vents, exhaust vents, etc., while improving the acoustic environment within a room.** Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

In 2011, Bonnie Schnitta of SoundSense received a patent for an Acoustic Muffler. This Muffler addresses the issue of significant sound transmission, or acoustic leakage, through openings in structures, such as a ceiling or wall, where airflow or heat dissipation is required. Examples of applications will be presented, such as openings created by recessed lights, speakers, or intake and exhaust vents. These openings in the structure significantly reduce the STC and/or IIC of the structure. The SoundSense Acoustic Muffler reduces acoustic leakage through an opening in a ceiling or wall that would potentially cause significant degradation in acoustic efficacy of the structure without resulting in any substantial pressure drop, while simultaneously allowing for the required air flow or heat dissipation.

The original application of the recessed light and additional similar applications as well as secondary purposes for this muffler will be detailed. One such secondary purpose is when the hole created by the recessed lights in the ceiling contributes to the room sounding better. The patented muffler not only allows this benefit to remain, but serves to inhibit the frequency(s) of concern from disturbing adjacent rooms. Cell tower equipment room noise reduction requiring airflow will also be detailed.

**9:25**

**2aAA5. A comparison of predicted total absorption of different sized and shaped materials using traditional "absorption coefficient" vs a proposed absorption constant.** Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

After using traditional absorption coefficients and calculating total absorptions for a room and then underestimating room RTs, it was decided that a new constant was needed that could be used to determine actual absorption of a surface. A constant was developed that meets this need and this paper shows the differences in the predicted absorptions using both methods vs using the new constant and the advantages of using the new constant.

## *Contributed Papers*

**9:45**

**2aAA6. Laboratory measurement of the acoustic absorption coefficient based on the modal dispersion.** Jevgenija Prisutova, Kirill Horoshenkov (Univ. of Bradford, Richmond Rd., Bradford BD7 1DP, United Kingdom, k.horoshenkov@bradford.ac.uk), Jean-Philippe Groby, and Bruno Brouard (Laboratoire d'Acoustique de l'Université du Maine, Université du Maine, Le Mans, France)

This work presents a novel method of measurement of the absorption coefficient of large material samples in an acoustic waveguide in a broad frequency range. The material sample is deployed at one end of an acoustic waveguide the other end of which is excited with a point source. The sound pressure data are obtained using a long horizontal microphone array deployed in this waveguide. The optimization analysis is then applied to the sound pressure data to calculate the modal reflection coefficients, which are then combined to determine the overall absorption coefficient of the material sample placed at the end of this waveguide. It believed that this method will be able to extend significantly the frequency range attained with the current ISO 10543-2 impedance tube method and be applied to those materials which have a corrugated surface or complex surface morphology such as acoustic diffusers or living plants. It is also believed that this method will provide the means to estimate efficiently the diffusivity of materials with complex surface morphology with a relatively simple laboratory setup.

**10:00–10:15 Break**

**10:15**

**2aAA7. Uncertainty of normal-incidence absorption coefficient measurements using the two-microphone cross-spectral method.** Matthew G. Blevins, Joshua Thede, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182, mgblevins@unomaha.edu)

Measured absorption coefficients have some uncertainty that can be expressed in terms of the uncertainty associated with each measured input quantity. The uncertainties of input quantities contribute to the combined uncertainty of the absorption coefficients in varying degrees dependent on the underlying relationship between each parameter. In this study, the propagation of uncertainty of the two-microphone cross-spectral method for measuring normal-incidence absorption coefficients is analyzed according to the ISO/IEC Guide 98-3:2008 "Guide to the expression of uncertainty in measurement." The results of an experimental investigation are explored to determine the chief sources of systematic error and the relationship between uncertainty of input quantities and uncertainty of intermediate calculations.

**10:30**

**2aAA8. Optimization of sound absorption performance of a new ecological material.** Seda Karabulut (MEZZO Studyo, Dept. of Architecture, Middle East Tech. Univ., ODTU Kosgeb Tekmer No112, Ankara 06800, Turkey, sedakarabulutt@gmail.com) and Mehmet Čalışkan (Dept. of Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

Material selection is foremost design parameter in providing acoustical comfort levels in both acoustically sensitive and noise generating spaces ranging from the small size meeting halls to large scale multipurpose auditoriums and even from foyers to shopping malls. Architects usually prefer seamless, unperforated materials in line with their interior design concepts. One of the objectives of this research is to contribute to the market portfolio of smooth faced, seamless acoustical materials with highest sound absorption performances. Another objective is to develop such a composition with ecological and sustainable ingredients, and binding techniques. Energy efficient and sustainable materials are frequently devised in construction industry for acoustically sensitive environments to get credits for international certification procedures such as LEED and BREEAM. Nevertheless, most of the acoustic materials in construction industry are perforated and/or supported with mineral wool based material backing, which have an adverse effect on indoor air quality. This article is on the improvement of an ecological unperforated sound absorptive material which is made of reed and pumice stone layers. The feasibility and effectiveness of a proposed configuration have already been studied. This paper seeks ways of optimizing number and thicknesses of different material layers in attaining maximum sound absorption performance.

**10:45**

**2aAA9. Acoustic determination of impedance tube microphone locations.** Cameron Fackler, Theodore S. Pitney, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, facklc@rpi.edu)

Impedance tube methods allow for convenient, rapid characterization of the normal-incidence acoustic properties of materials and constructions. Many such methods rely on the determination of precise phase relationships between microphones sensing the sound field inside the impedance tube; knowledge of the location of these microphones is crucial to the accuracy of the measurement. Due to the large physical size of typical microphone diaphragms, physical measurements of the microphone positions (such as with a ruler or caliper) are inadequate and have a large uncertainty. This paper presents a method to determine the impedance tube microphone acoustic center locations from a broadband acoustic reflection coefficient measurement of a rigid termination. Utilizing the Bayesian inference framework, the estimation procedure provides information about microphone locations and uncertainties in the position estimates.

11:00

**2aAA10. Detection of specular and diffuse reflections in concert halls using continuous wavelet transforms.** Jin Yong Jeon and Muhammad Imran (Dept. of Architectural Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul, 133791, South Korea, jyjeon@hanyang.ac.kr)

Specular and diffuse reflections have been detected in fine spatio-temporal structures of room impulse responses (IRs) at different positions in a 1:10 scale model hall. Continuous wavelet transformation (CWT) coefficients were calculated using mother wavelet functions of the Daubechies wavelet families and specular reflections along with their degrees of diffuseness were investigated from the IRs. In CWT analysis, the early specular reflections were detected by fining their similarities with mother wavelet at different scales. While these reflections were treated as "defects" or "singularities" that embedded in the Schroeder decay curve. In addition, the time difference of arrival (TDOA) was applied to localize the reflections using a cross correlation function for time delay estimation and direction finding of binaural room IRs. The spectral characteristics in terms of delay time were calculated by auto-correlation and interaural cross-correlation functions (ACF/IACF).

11:15

**2aAA11. Bayesian parameter estimation of single- and double-layer micro-perforated panel absorbers.** Andrew Schmitt, Ning Xiang, and Cameron Fackler (Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, aajschmitt@gmail.com)

Micro-perforated panel (MPP) absorbers have been widely accepted in applications as an efficient and clean sound-absorption solution providing extremely high, yet relatively narrow frequency band, acoustic absorption coefficients. Design for the custom acoustic performance of MPPs can be challenging for engineers and designers, however, due to the number of parameters making up the panel absorbers. This work examines the effects of the perforation rate, pore diameter, airspace depth, and panel thickness on the absorption profile of single- and double-layer MPP absorbers. A Bayesian inference framework is implemented in order to inversely determine parameter values and their uncertainties from experimentally measured acoustic absorption data of various MPP constructions of unknown parameters. This analysis also provides insight into the effects of each parameter on the acoustic performance as well as interdependence among the parameters.

11:30

**2aAA12. Acoustics 101 for architects.** Michael W. Fay (Contracting Div., Sound Image, 2415 Auto Park Way, Escondido, CA 92029, mfay@sound-image.com)

A presentation of acoustical terminology and concepts that relate directly to the design and construction of an architectural space, non-technical explanations, descriptions, and examples. Architectural acoustics can be defined as the study and application of acoustic principles as they are applied to the inside of a building or structure. Usually, these are buildings that will be used for a live performance or the presentation of other useful information. This paper is written with the architect in mind; to help define and explain some of the most basic concepts relating to architectural acoustics. In today's pro/commercial audio marketplace, architectural acoustics must play a significant role in the design process for any programmed meeting or entertainment space. This work puts essential terms and concepts into the hands of the architect, owner, or anyone else who would like to have a better understanding of this topic. Contents: What is Sound? Sound Propagation Three Acoustical Tools Where Does All the Unused Sound Go After it's Been Heard? Audio Volume Changes—What Do the Numbers Mean? The Inverse Square Law Room Geometry—The Good, Bad, and Ugly Reverberation and Echo Speech Intelligibility Noise Internal vs External Noise Room Modes Variable Acoustics Psychoacoustics Conclusion.



TUESDAY MORNING, 3 DECEMBER 2013 UNION SQUARE 23/24, 7:55 A.M. TO 11:50 A.M.

## Session 2aAB

## Animal Bioacoustics and Acoustical Oceanography: Broadening Applications of Tags to Study Animal Bioacoustics I

Marla M. Holt, Cochair
*NOAA NMFS NWFSC, 2725 Montlake Blvd. East, Seattle, WA 98112*

Alison K. Stimpert, Cochair
*Dept. of Oceanogr., Naval Postgraduate School, Monterey, CA 93943*

**Chair's Introduction—7:55**

### *Invited Papers*

8:00

**2aAB1. History of applications of tags to study animal bioacoustics.** Peter L. Tyack (Biology, Univ. of St Andrews, Sea Mammal Res. Unit, Scottish Oceans Inst., St Andrews, Fife KY16 8LB, United Kingdom, plt@st-andrews.ac.uk)

I discuss the history of applications of tags to study animal bioacoustics, with an emphasis on toothed whales. My own interest in this topic was stimulated in the early 1980s by problems identifying which dolphin made a sound. Our inability to identify signaler and receiver hindered the study of communication among marine mammals. I discuss the evolution of devices and methods to solve this problem. The development of acoustic recording tags in the 1990s enabled the capacity to monitor what an animal hears at sea and how it responds. These tags form a critical enabler for field experiments on the relationship between acoustic exposure measured on the tag and behavioral responses, also often measured by the tag. Use of passive acoustic tags with echolocating animals has opened a new window on how toothed whales echolocate to find, approach, and capture prey, especially when the tags also include three-axis accelerometry and magnetometry to measure orientation and movement. The combination of these tags with passive acoustic monitoring provides a powerful method to improve localization, to estimate the three dimensional beampattern of sound production, and to estimate the absolute abundance of species that vocalize.

**8:20**

**2aAB2. Acoustic time synchronization among tags on porpoises to observe their social relationships.** Tomonari Akamatsu (Res. Ctr. for Fisheries System Eng., Fisheries Res. Agency & JST CREST, 7620-7, Hasaki, Kamisu, Ibaraki 314-0408, Japan, akamatsu@affrc.go.jp), Mai Sakai (Wildlife Res. Ctr., Univ. of Kyoto & JSPS, Kyoto, Japan), Ding Wang, Kexiong Wang, and Songhai Li (Key Lab. of Aquatic Biodiversity and Conservation of the Chinese Acad. of Sci., Inst. of Hydrobiology of the Chinese Acad. of Sci., Wuhan, China)

Observing and monitoring underwater social interactions of cetaceans is challenging. Because cetaceans spend most of their time underwater, it is important to monitor their underwater behavior individually. The finless porpoise is small and has no available natural identification marks that causes little knowledge of its sociality. Here we used acoustic datalogger to synchronize individual depth profile among individuals within a second. Acoustic and behavior tags were deployed on six free-ranging finless porpoises simultaneously and released in open water. Echolocation sounds were used as the trigger signal to synchronize the clock of all logging systems. Synchronous dives characterized by similar time-depth profile were used as an index of association. Two pairs tended to participate in long periods of synchronized diving more frequently than 13 other possible pairs, indicating that these four porpoises chose their social partners. Initiator and follower could be identified by precisely time synchronized data. The adult males tended to follow the immature female and juvenile male, respectively. However, the role of an initiator often changed within the pair during synchronized diving, and their body movements appeared to be non-agonistic. The time-synchronized bio-logging method was useful for observation of the social relationships of free-ranging aquatic animals.

**8:40**

**2aAB3. Studying acoustic communication in pilot whale social groups.** Frants H. Jensen (Biology, Woods Hole Oceanographic Inst., 266 Woods Hole Rd., M.S. # 50, Woods Hole, MA 02543, frants.jensen@gmail.com) and Peter L. Tyack (Scottish Oceans Inst., Univ. of St Andrews, St Andrews, United Kingdom)

Many cetaceans are gregarious animals with a complex group structure, and they depend on acoustic signals for mediating social interactions among individuals. However, the marine lifestyle and closed sound production system makes it difficult to study social signaling in groups of wild cetaceans. Acoustic and movement logging tags offer new possibilities for sampling the sounds and behavior of individuals, but themselves provide new challenges in determining the source of acoustic signals. Here, we draw on experiences from studies of short-finned and long-finned pilot whales to discuss how social signaling can be investigated in wild marine mammals. We discuss how specific social contexts, especially separations from the social group, can aid the interpretation of individual tag data to test whether calls of short-finned pilot whales are important in mediating social contact with group members, while emphasizing the pitfalls of using such methods. Specifically, we highlight the advantages of simultaneously instrumenting multiple closely associated pilot whales with acoustic and movement recording tags. This has improved our understanding of acoustic interactions through ready identification of the sender and simultaneously monitoring the reaction of other group members, and we use this dataset to discuss ongoing challenges of studying social dynamics using simultaneous tag deployments.

**9:00**

**2aAB4. Insights into a complex communication system from tagged bottlenose dolphins.** Laela Sayigh (Biology Dept., Woods Hole Oceanographic Inst., M.S. #50, Woods Hole, MA 02543, lsayigh@whoi.edu), Vincent Janik (Biology Dept., Univ. of St. Andrews, St. Andrews, United Kingdom), Frants Jensen (Biology Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Katherine McHugh, Randall Wells (Sarasota Dolphin Res. Program, Chicago Zoological Society, c/o Mote Marine Lab., Sarasota, FL), and Peter Tyack (Biology Dept., Univ. of St. Andrews, St. Andrews, United Kingdom)

Since 2011, we have deployed 30 acoustic and movement logging DTAGs on long-term, multi-generational resident bottlenose dolphins in Sarasota Bay, Florida, for a total of approximately 140 h. Twenty-two tags were deployed simultaneously on pairs of associated individuals, allowing for greater resolution of individual vocal activity. Virtually all dolphins in the Sarasota Bay community are identifiable both visually and by means of their individually distinctive signature whistles. Tags were attached during brief capture-release health assessments, and behavioral observations of tagged individuals post-release continued for as long as possible. Tag data reveal unique insights into foraging behavior, including distinctive acoustic and movement patterns associated with particular foraging modes (e.g., “pinwheel feeding”). In addition to echolocation clicks and buzzes, several distinctive pulsed sounds were recorded on the tags. Whistle copying was observed 18 times in a preliminary analysis of approximately two hours of data, and at least one instance involved more than two dolphins producing the same whistle. Finally, we obtained evidence for at least one shared, stereotyped non-signature whistle. Combining extensive longitudinal information on individual dolphins with fine scale behavioral and acoustic data provides tremendous opportunities for describing and quantifying the complexity of the bottlenose dolphin communication system.

**9:20**

**2aAB5. Challenges in identifying (or not) focal animal sound production in baleen whale acoustic tag datasets.** Alison K. Stimpert (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943, alison.stimpert@gmail.com), Doug P. Nowacek (Nicholas School, Duke Univ., Beaufort, NC), Ari S. Friedlaender (Southall Environ. Associates, Inc., Aptos, CA), Jan Straley (Biology, Univ. of Alaska Southeast, Sitka, AK), David W. Johnston (Nicholas School, Duke Univ., Beaufort, NC), Jeremy A. Goldbogen (Cascadia Res. Collective, Olympia, WA), and Ching-Sang Chiu (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA)

Ascribing sounds on animal-borne tag recordings to individual sound producers is integral to understanding social behavior of animal groups. Previously, sounds recorded on tags have been assigned to the tagged individual (focal animal) based on proximity of other conspecifics, angle of arrival, low frequency artifacts in the sound, or a combination of signal-to-noise ratio (SNR) and received level (RL). However, most acoustic-based methods do not translate well to baleen whales producing low frequency sounds, as the tag often resides in the near field of the sound source. In addition, for social species that spend time in groups with conspecifics in close proximity,

sounds produced by nearby animals may have comparably high SNR and RL. Here we discuss the challenges of determining if a tagged whale is calling in baleen whale datasets, using acoustic records from two humpback whales, one fin whale, and one blue whale as examples. The datasets include intense song or feeding calls and are from several locations. We compare SNR, RL, harmonic content, and behavioral sensor data in these cases, and discuss the implications of confirming sound production by a tagged individual for measuring communication, behavior, and responses to external stimuli in baleen whales.

**9:40**

**2aAB6. Tags, drifters, and Towfish: Using multiple recording platforms to characterize odontocete acoustic space.** T. A. Mooney, Maxwell B. Kaplan (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, amooney@whoi.edu), Robin W. Baird (Cascadia Res. Collective, Olympia, WA), and Jim Partan (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., Woods Hole, MA)

Bioacoustic tags can reveal novel information about the behavior and ecology of animals on which they are deployed. Yet tags are often placed off the animals' acoustic axis, limiting some potential analyses. In order to broaden abilities to examine bioacoustic signals and behavior of several Hawaiian odontocetes we adapted recording methods to enhance data collection opportunities and free-field records. While bioacoustic DTAGs were deployed, we also used DMONs (digital acoustic recorders) in both a GPS-outfitted drifter buoy (Drifting Acoustic Wideband Gizmo = DAWG) and a Towfish around pantropical spotted dolphins (*Sa*), melon-headed whales (*Pe*), and short-finned pilot whales (*Gm*). Daytime tag recordings show *Pe* and *Sa* were limited to relatively shallow dives ($< 50$ m) but were relatively soniferous, whereas *Gm* made occasional deeper dives (to 700 m) and fewer individual calls. Group measures for *Pe* and *Sa* from the DAWG and Towfish revealed relatively high incidences of overlapping calls. Preliminary investigations of *Pe* whistles suggest some limited variation between *Pe* populations and considerable variability in individual call types. Such characterizations of call rates and variability support efforts to detect and classify odontocete calls. The different methods provided complementary means to collect substantial bioacoustic data on pelagic odontocetes for which few data exist.



**10:00–10:15 Break**

**10:15**

**2aAB7. Classification of behavioral state using hidden Markov model analysis of animal-attached tag data: Applications and future prospects.** Patrick J. Miller and Saana Isojunno (School of Biology, Univ. of St Andrews, Bute Bldg., St Andrews, Fife KY16 9QQ, United Kingdom, pm29@st-andrews.ac.uk)

Data from high-resolution animal-attached tags enable quantification of behavioral responses to anthropogenic noise. However, the duration of such detailed tag records on marine divers are typically too short to allow evaluation of the biological significance of such effects. To explore whether and how sperm whale behavior changed during exposure to sonar, we developed a discrete-time hidden activity state model that describes how observed parameters derived from measured Dtag data (depth, pitch, and clicking behavior) arise from five behavioral modes (surfacing, descent, bottom phase, ascent, resting, and silent active). Although the model assumed simple Markovian state-transitions, the state classification matched well with expert judgment of dive state. During experimental exposures to 1–2 kHz sonar, all four sperm whales tested reduced foraging time, increased silent active behavior, and buzz rates during foraging states decreased. None of those effects were found during 6–7 kHz experimental exposures of the same four whales, nor for three other whales exposed to distant sonar. Hidden classification of behavioral state using quantitative analysis of data collected by the animal attached tag is a procedure that has the potential to be processed autonomously on-board tags. This would enable collection and satellite telemetry of longer-term behavioral data sets with biologically significant interpretations.

**10:35**

**2aAB8. Statistical analysis of data from acoustic tags: Methods for combining data streams and modeling animal behavior.** Stacy L. DeRuiter, Catriona Harris, Dina Sadykova, and Len Thomas (School of Mathematics and Statistics, Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, CREEM, St. Andrews KY169LZ, United Kingdom, sldr@st-andrews.ac.uk)

Statistical analysis of data from multi-sensor acoustic tags presents several characteristic challenges. Datasets generally include time-series of many measurements on a small number of individuals; different data streams often have distinct temporal resolutions and precisions. The MOCHA project (Multi-study Ocean acoustics Human effects Analysis) is a three-year effort focused on developing innovative statistical methods for such data. Here, we present several approaches for appropriate, effective statistical analysis of such datasets, with an emphasis on quantitative assessment of changes in marine mammal behavior in response to acoustic disturbance. Issues to be addressed will include: combining data streams from multi-sensor tags (and also concurrent visual observation data) for statistical analysis; statistical methods to characterize or summarize normal behavior and detect departures from normal; methods for analysis of call-production-rate data from acoustic tags; and methods for combining analysis of data from multiple tags, individuals, and species. Specific statistical methods to be presented will include Mahalanobis distance as a summary of multivariate data, state-switching models, random effects, and other extensions of generalized linear models appropriate to tag data.

**10:55**

**2aAB9. Using acoustic tags to investigate sound exposure and effects on behavior in endangered killer whales (*Orcinus orca*).** Marla M. Holt, M. Bradley Hanson, Candice K. Emmons (Conservation Biology Div., NOAA NMFS NWFSC, 2725 Montlake Blvd. East, Seattle, WA 98112, Marla.Holt@noaa.gov), Juliana Houghton (School of Aquatic and Fishery Sci., Univ. of Washington, Seattle, WA), Deborah Giles (Dept. of Wildlife, Fish and Conservation Biology, Univ. of California, Davis, Seattle, CA), Robin W. Baird, and Jeff Hogan (Cascadia Res. Collective, Olympia, WA)

In this investigation, acoustic tags (DTAGs) allow us to better understand noise exposure and potential behavioral effects in endangered Southern Resident killer whales (SRKWs). Designated critical habit of SRKWs includes summer foraging areas where vessel traffic from commercial shipping, whale-watching, and other boating activities is common. Risk factors of population recovery include

vessel and noise effects, and prey quality and availability. DTAGs, equipped with hydrophones and other sensors, are attached to individual whales to collect data on vocal and movement behavior, as well as their acoustic environment. Specific research goals include: (1) quantifying received noise levels in biologically relevant frequency ranges from tag data; (2) determining relationships between noise levels and detailed vessel variables obtained from precise geo-referenced vessel data collected concurrently; (3) investigating whale acoustic and movement behavior during different activities, including foraging, to understand sound use and behavior in specific biological and environmental contexts; and (4) determining potential effects of vessels and associated noise on behavior. This paper will describe the experimental approach taken, challenges faced, and results obtained from over 80 h of tag data. These data have been critical for addressing our research goals related to multiple population risk factors of endangered SRKWs.

**11:15**

**2aAB10. Killers in the dark—Acoustic evidence for night-time predation by mammal-eating killer whales (*Orcinus orca*) in Alaska.** Volker B. Deecke (Ctr. for Wildlife Conservation, Univ. of Cumbria, Penrith CA11 0AH, United Kingdom, volker.deecke@cumbria.ac.uk), Ari D. Shapiro (Biology Dept., Woods Hole Oceanographic Inst., Boston, MA), and Patrick J. Miller (Sea Mammal Res. Unit, Univ. of St. Andrews, St. Andrews, United Kingdom)

North Pacific killer whales that specialize on hunting marine mammals do not typically echolocate while searching for prey. This suggests that they detect prey by either relying on visual cues or listening for sounds generated by prey animals. If prey detection requires vision, hunting should be limited to the daylight hours. Documenting predation at night would therefore provide evidence supporting a passive listening hypothesis of prey detection. We used digital recording tags (DTAGs) to study the behavior of mammal-eating killer whales in Southeast Alaska. These tags recorded the underwater movements of the tagged individual and any sound emitted or received. Predation events were identified using distinctive sounds generated during prey capture and handling. We deployed 13 tags, of which 7 remained attached for at least part of the night. The majority of tags recorded night-time predation, even though nights were short (average of 4:18 h) during the study period. These findings show that mammal-eating killer whales can detect prey at night and thus suggest that passive listening is an important part of their hunting strategy. Acoustic data from digital recording tags can therefore provide valuable insights into the night-time activities and foraging behavior of killer whales and other marine mammals.

## *Contributed Paper*

**11:35**

**2aAB11. Acoustic and foraging behavior of tagged sperm whales under natural and depredation foraging conditions in the Gulf of Alaska.** Delphine Mathias (GIPSA-Lab, 11 rue des Mathématiques, Saint Martin d'Hères 38402, France, delphine.mathias@gmail.com), Lauren Wild (Sitka Sound Sci. Ctr., Sitka, AK), Aaron Thode (Scripps Inst. of Oceanogr., La Jolla, CA), Jan Straley (Univ. of Alaska Southeast, Sitka, AK), John Calambokidis, and Greg S. Schorr (Cascadia Res. Collective, Olympia, WA)

Sperm whales have been depredating black cod (*Anoplopoma fimbria*) from demersal longlines in the Gulf of Alaska for decades, but the behavior has now become pervasive enough that it may be affecting government estimates of the sustainable catch, motivating further studies of this behavior. Over a three-year period, 11 B-Probe bioacoustic tags have been attached to seven adult sperm whales off Southeast Alaska, permitting observations of the animals' dive profiles and acoustic behavior during natural and depredation foraging conditions. Two rough categories of depredation were identified: "deep" and "shallow." "Deep depredating" whales consistently surface within 500 m of a hauling fishing vessel, have maximum dive depths greater than 200m, and display significantly different acoustic behavior than naturally foraging whales, with shorter inter-click intervals, occasional bouts of high "creak" rates, and fewer dives without creaks. "Shallow depredating" whales conduct dives that are much shorter, shallower, and more acoustically active than both the natural and deep depredating behaviors, with median creak rates three times that of natural levels. Occurrence of slow clicks and the behavioral context in which these vocalizations are produced were also investigated. These results provide insight into the energetic benefits of depredation behavior to sperm whales. [Work conducted under the SEAS-WAP program, supported by the North Pacific Research Board and the National Geographic Society.]

# Session 2aBA

## Biomedical Acoustics and Signal Processing in Acoustics: Application of Acoustic Radiation Force in Medical Imaging

Mostafa Fatemi, Chair
*Physiol. & Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905*

**Chair's Introduction—7:55**



### *Invited Papers*

**8:00**

**2aBA1. Acoustic radiation force on gas bubbles and soft elastic scatterers in tissue.** Sangpil Yoon, Salavat R. Aglyamov, Andrei B. Karpiouk (Dept. of Biomedical Eng., The Univ. of Texas at Austin, Austin, TX), Stanislav Y. Emelianov (Dept. of Biomedical Eng., The Univ. of Texas at Austin, Austin, Tennessee), Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Acoustic radiation force on a scatterer in tissue depends on the compressibility and shear modulus of both the tissue and the scatterer. This force is related to the monopole and dipole scattering coefficients. The finite shear modulus of the tissue decreases the radiation force in comparison with the force exerted on the same scatterer surrounded by liquid. Shear moduli for soft tissue range from several kilopascals (breast, liver) to tens of kilopascals and higher for cornea, cartilage, and cancerous tissue. As reported previously, the radiation force on a bubble in tissue having 100 kPa shear modulus is 50% less than if the bubble is in water. This difference decreases for scatterers with finite shear moduli, examples of which are reported here. Additionally, displacement of a scatterer due to radiation force is inversely proportional to the shear modulus of the tissue, which permits measurement of the latter. Experiments demonstrating this technique are reviewed. In these experiments, the radiation force is applied to a gas microbubble produced by laser-induced optical breakdown, while displacement of the microbubble is measured by high-frequency ultrasound as a function of time. Results are reported for tissue-mimicking phantoms and animal crystalline lenses *in vitro*.

**8:20**

**2aBA2. A review of the medical applications of shear wave elastography.** Mickael Tanter, Mathieu Pernot, Jean Luc Gennisson, and Mathias Fink (Langevin Inst., ESPCI, 1 rue jussieu, Paris 75005, France, mickael.tanter@espci.fr)

Supersonic shear wave elastography (SWE) is a quantitative stiffness imaging technique based on the combination of a radiation force induced in tissue by an ultrasonic beam and ultrafast ultrasound imaging sequence (up to more than 10,000 frames per second) catching in real time the propagation of the resulting shear waves. Local shear wave speed is estimated and enables the two dimensional mapping of shear elasticity. This imaging modality is implemented on conventional probes driven by dedicated ultrafast echographic devices and can be performed during a standard ultrasound exam. The clinical potential of SSI is today extensively investigated for many potential applications such as breast cancer diagnosis, liver fibrosis staging, cardiovascular applications, and ophthalmology. This invited lecture will present an overview of the current investigated applications of SSI and the new trends of shear wave elastography research topics.

**8:40**

**2aBA3. Application of shear wave imaging and shear wave dispersion ultrasound vibrometry in assessing viscoelastic properties of human thyroid: *In vivo* pilot study.** Mohammad Mehrmohammadi, Pengfei Song, Carolina A. Carrascal, Matthew W. Urban (Physiol. and Biomedical Eng., Mayo Clinic, 200 First St. SW, Rochester, MN 55905, mehrmohammadi.mohammad@mayo.edu), Matthew R. Callstrom (Radiology-Diagnostic, Mayo Clinic, Rochester, MN), John C. Morris (Endocrinology, Mayo Clinic, Rochester, MN), Shigao Chen, James F. Greenleaf, Mostafa Fatemi, and Azra Alizad (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Thyroid cancer is the fastest growing age and gender adjusted cancer in 2011 according the American Cancer Society. The majority of the clinically diagnosed thyroid nodules are benign while less than 5% represent intrathyroidal cancers. Currently, the clinical gold-standard procedure for assessing the thyroid nodules is needle biopsy, a procedure that is associated with significant financial burden as well as pain and risk for patients. Therefore, a noninvasive, affordable, and potentially widely available method to differentiate between benign and malignant thyroid nodules can play an important role in reducing the number of unnecessary biopsies. In this study, we investigate the feasibility of two acoustic radiation force elastography techniques, shear wave dispersion ultrasound vibrometry (SDUV) and comb-push ultrasound shear wave elastography (CUSE imaging), in identifying thyroid nodules (imaging) and differentiating between benign and malignant pathologies based on their elasticity and viscosity (SDUV measurements). Our preliminary results show that the measured shear elasticity and shear viscosity parameters depend on tissue type; hence, these measurements may be utilized to differentiate between healthy normal thyroid tissue, benign nodules, and malignant nodules. Further studies on a large population of patients is required to better evaluate the role of the combination of elasticity and viscosity properties of tissue in differentiating various thyroid nodules.

## *Contributed Papers*

**9:00**

**2aBA4. Ultrasound-based shear wave evaluation in transverse isotropic tissue mimicking phantoms.** Sara Aristizabal, Randall R. Kinnick, Carolina Amador, Ivan Z. Nenadic, James F. Greenleaf, and Matthew W. Urban (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 1st St. SW, Rochester, MN 55905, aristizabaltaborda.sara@mayo.edu)

Introduction: Ultrasound radiation force-based methods can quantitatively evaluate tissue viscoelastic material properties. A limitation of current methods includes neglecting the inherent anisotropy nature of tissue. To explore this phenomenon, we created a phantom incorporating fibrous material that has preferential orientations. Methods: Two phantoms were made in a cube-shaped mold using a fibrous material arranged in multiple layers and embedded in porcine gelatin using two different concentrations of the gelatin (8%, 14%). Shear wave measurements were made in the phantoms at different angles by rotating the phantom, where 0° and 180° were defined along the fibers, and 90° and 270° across the fibers. Measurements were performed using a Verasonics ultrasound system equipped with a linear array transducer. Results/Discussion: The mean shear wave speeds and mean standard deviations for 8% and 14% gelatin along the fibers (0°) were (3.60 ± 0.03 and 4.10 ± 0.11 m/s) and across the fibers (90°) were (3.18 ± 0.12 and 3.90 ± 0.02 m/s), respectively. Conclusion: The fibrous gelatin-based phantoms exhibited anisotropy that could be measured using quantitative shear waves speed measurements. Increasing the gelatin percentage increases the shear wave speed and anisotropic moduli. [This study was supported by NIH grant DK092255.]

**9:15**

**2aBA5. Characterizing dynamics of shear waves induced with acoustic radiation force impulse in histotripsy lesions for treatment feedback.** Tzu-Yin Wang (Radiology, Stanford Univ., 1201 Welch Rd., Stanford, CA 94305, tzuyin@stanford.edu), Timothy L. Hall, Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), J. Brian Fowlkes (Radiology, Univ. of Michigan, Ann Arbor, MI), and Charles A. Cain (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Histotripsy mechanically fractionates soft tissues into fluid-like homogenates that cannot support shear waves. We hypothesize that dynamics of shear waves excited from a histotripsy lesion using acoustic radiation force impulse (ARFI) change progressively during the fractionation process, and such change is related to the degree of tissue fractionation. To test this hypothesis, lesions with different degrees of fractionation were created in agar-graphite tissue mimicking phantoms and *ex vivo* kidneys with increasing numbers of histotripsy pulses (3-cycle 750- kHz ultrasound pulses at a peak negative/positive pressure of 17/108 MPa). The shear waves were excited by ARFI focused at the lesion center. The shear-induced temporal displacement profile was measured at a lateral location 10 mm offset to the lesion with M-mode imaging. Results showed significant changes in two characteristics: the peak-to-peak displacement decayed exponentially, and the relative time-to-peak displacement increased and saturated with increasing numbers of histotripsy pulses ($N=6$). Correspondingly, the degree of tissues fractionation, as indicated by the percentage of structurally intact cell nuclei, decreased exponentially. Strong linear correlations existed between the two characteristics and the degree of tissue fractionation ($R2 > 0.97$). These results suggest that characteristics of shear waves induced in a histotripsy lesion may provide useful feedback for treatment outcomes.

**9:30**

**2aBA6. Robust shear wave motion tracking using ultrasound harmonic imaging.** Pengfei Song, Heng Zhao, Matthew W. Urban, Armando Manduca (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Med. Sci. 1-24, 200 First St. SW, Rochester, MN 55905, song.pengfei@mayo.edu), Sorin V. Pislaru (Cardiovascular Diseases, Mayo Clinic College of Medicine, Rochester, MN), Randall R. Kinnick, James F. Greenleaf, and Shigao Chen (Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN)

Robust shear-wave motion detection is essential for producing reliable shear elasticity measurements for ultrasound shear-wave elastography (SWE). However, because shear-wave motion is extracted from radiofrequency signals which can be heavily contaminated by noise, shear-wave motion detection can be very challenging, especially for *in vivo* applications. This study investigated the implementation of harmonic imaging (HI) to facilitate more robust shear-wave tracking based on HI's effectiveness in suppressing imaging noise associated with ultrasound reverberation, phase aberration, and clutter noise. A HI shear-wave tracking sequence was developed combining the pulse-inversion HI method with the plane wave imaging technique to transmit phase-inverted pulses at a high frame rate of several kilohertz. The backscattered ultrasound signals from phase-inverted pulses were added to suppress the fundamental and enhance the second harmonic component, from which the shear-wave motion was extracted. A pork belly phantom experiment showed that HI could significantly improve shear-wave motion detection by producing almost three-fold less underestimation of shear-wave motion and over ten-fold more precision for shear-wave speed measurements than fundamental imaging. An *in vivo* transthoracic case study of a human heart showed that HI substantially improved the success rate of shear-wave motion detection and could provide consistent estimates of the left ventricle myocardium stiffness.

**9:45**

**2aBA7. Multipoint measurement of sound pressure and temperature in biological tissues by using optical fiber sensors.** Takashi Kageyama (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, dmm1013@mail4.doshisha.ac.jp), Daisuke Koyama (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), and Iwaki Akiyama (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

Acoustic radiation force impulse (ARFI) has been actively studied in the field of medical ultrasonics. ARFI generates shear wave in the tissue to evaluate the tissue hardness using that velocity, and this method is applied to the diagnosis of liver tumor. Generally, long pulses are used for ARFI; therefore, we need to take account of thermal and physical effects on biological body. According to the regulation of Food and Drug Administration (FDA), the acoustic output of diagnostic ultrasound is approved as follows: $I_{spta.3} < 720$ mW/cm$^2$, MI $< 1.9$. However, there are some reports that we have possibilities to affect body thermally and physically even under these conditions. In this report, we propose the optical fiber system using fiber Bragg grating (FBG) to measure ultrasound pressure and temperature change separately. The purpose is that we measure the temperature range of 4 degrees at the resolution of 0.1 degree and the sound pressure range of 10 MPa. We utilized an optical fiber which had 10 mm FBG in the center; moreover a narrowband light source and a photo detector to modulate reflected light signals by FBG to electrical signals. We obtained 3.2 $\mu$V/Pa and 9.5 pm/degree as the sensitivities of 10 mm FBG.

**10:00**

**2aBA8. Angle exchange symmetries for acoustic Bessel beam scattering and the radiation force on spheres.** Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Bessel beams are characterized by a cone angle $\beta$ and as previously noted there is an exchange symmetry in the scattering involving the cone and the scattering angle $\theta$ [Marston, J. Acoust. Soc. Am. **121**, 753–758 (2007)]. The sphere is taken to be on the beam's axis. In the present study a broader class of exchange symmetries in the scattering pattern in the ($\theta$, $\beta$) domain are noted for vortex as well as zero-order beams. Reflection exchange symmetries are present in the scattering pattern about the lines: $\theta = \beta$ and $(180° - \theta) = \beta$. The radiation force is known to be affected by the angular asymmetry of the scattering pattern as a function of scattering angle [Zhang and Marston, Phys. Rev. E **84**, 035601 (2011)]. It follows from angle exchange that the radiation force is similarly related to cone-angle asymmetry of the scattering pattern for a fixed scattering angle. This applies to negative as well as positive forces and is easily illustrated. [Work supported by ONR.]

**10:15–10:30 Break**

10:30

**2aBA9. Evaluation of frequency characteristics of shear waves produced by unfocused and focused beams.** Matthew W. Urban, Carolina Amador, Heng Zhao (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), Mingu Lee, Hwan Shim (Samsung Electronics Co., Suwon-Si, South Korea), Shigao Chen, and James F. Greenleaf (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN)

Measurements of viscoelasticity with shear wave velocity dispersion requires measurements over a large bandwidth. In this study, we explored the parameters that modulate the frequency characteristics of shear waves induced using radiation force push beams. We used a Verasonics ultrasound scanner equipped with a linear array transducer. We performed measurements of shear wave motion induced using both focused and unfocused ultrasound beams. Measurements were made in elastic phantoms with shear moduli of 1, 4, and 16 kPa. The number of elements used for the unfocused beams were varied from 8 to 24, and for the focused beams from 16 to 128. The shear wave motion was tracked using plane wave imaging, and a one--dimensional autocorrelation algorithm applied to the acquired in-phase/quadrature data. At each pixel we calculated the fast Fourier transform of the data and found the center frequency, center-of-gravity, and $-3$ dB bandwidth. We compared the frequency characteristics from the different push beams. The frequency characteristics were found to be spatially variant and dependent on the number of elements used as well as the shear modulus of the medium. The center frequency, center-of-gravity, and the bandwidth were found to be correlated to one another, and strongly associated with the stiffness of the medium.

10:45

**2aBA10. Model-free quantification of shear wave velocity and attenuation in tissues and its *in vivo* application.** Ivan Nenadic, Matthew W. Urban, Bo Qiang, Shigao Chen, and James Greenleaf (Mayo Clinic, 200 1st St. SW, Rochester, MN 55906, ivandulan@gmail.com)

We validate a technique for model-free measurement of shear wave velocity and attenuation. A mechanical shaker was used to excite harmonic plane and cylindrical waves in phantoms and excised tissue. Radiation force was used to excite impulsive cylindrical waves. 2D FFT of the displacement yielded the k-space whose coordinates are frequency and the wave number. The shear wave velocity at each frequency was obtained by finding the maximum at the given frequency in k-space and dividing the frequency coordinate by the wave number coordinate. The attenuation ($\alpha$) at a given frequency was calculated using $\alpha = \text{FWHM} \times \pi/\sqrt{3}$, where FWHM is the full width at half maximum of the k-space peak along the given frequency. This method was applied to measure shear wave velocity and attenuation of transplanted kidneys and livers, and in the thyroid tumor, and compare it to the healthy tissues. The velocities and attenuations at each frequency for various excitation methods agree within one standard deviation. The k-space estimates of velocity and attenuation agreed with those obtained using the phase gradient (velocity) and amplitude decay (attenuation). The transplanted organs and the thyroid tumor had higher velocity and lower attenuation than healthy tissues.

11:00

**2aBA11. Using equilibrium equation constraints to obtain precise lateral displacement estimates in ultrasound imaging.** Olalekan A. Babaniyi, Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, lekanb@bu.edu), and Assad A. Oberai (Mech., Aerosp., and Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY)

Ultrasound elasticity imaging, whether based on radiation force, quasistatic deformation, or other means, depends upon using ultrasound to measure tissue motion. Ultrasound image data with standard beamforming can provide very precise measurements of soft tissue displacement in the axial direction, i.e., in the direction of the ultrasound beam. Lateral (and elevational) displacement estimates are relatively noisy. The authors describe a new processing method designed to estimate a precise and accurate 2D full displacement vector field from accurate measurements of a single component, and a noisy measurement of a second component. The proposed variational approach finds the displacement field that best fits the data, but that satisfies Navier's equation locally. The equilibrium equation is automatically relaxed along internal interfaces in the material, which need not be identified beforehand. In this way, the method accommodates piecewise constant material property distributions without knowing in advance where the properties change. The iterative implementation for plane stress incompressible elasticity converges in 3–10 iterations. [Authors gratefully acknowledge funding from NSF and NIH (NSF Grant No. 50201109; NIH NCI-R01CA140271).]

11:15

**2aBA12. Approximate analytical models for cylindrical shear wave propagation in viscoelastic media.** Yiqun Yang (Elec. and Comput. Eng., Michigan State Univ., 2120 Eng. Bldg., East Lansing, MI 48864, mcgough@egr.msu.edu), Matthew W. Urban, Bob Qiang (Biomedical Eng., Mayo Clinic, Rochester, MN), and Robert J. McGough (Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Improved approximate frequency domain expressions are derived for cylindrical shear wave propagation in viscoelastic media. These expressions extend prior results that describe cylindrical wave propagation in lossless media. Previously, an analytical expression for a cylindrical wave was obtained in terms of a Hankel function and a large argument approximation was applied to the result. A leading frequency-dependent term was then treated as a constant with respect to frequency. In the improved expression, the frequency-dependence of the leading term is retained. For lossless media, the leading term is a fractional integrator, and for viscoelastic media, the leading term is either a fractional integrator or an integer-order integrator, depending on the frequency range. The lossless and the viscoelastic models are evaluated in the frequency domain for simplified source geometries and compared to numerical results. The comparison shows that the agreement between the analytical and the numerical models is excellent. Implications for time-domain calculations in viscous media will also be discussed. [This work was supported in part by NIH Grant Nos. R01 EB012079 and R01 DK092255.]

11:30

**2aBA13. Ultrasound bladder vibrometry for evaluation of bladder compliance: Preliminary *in vivo* results.** Mohammad Mehrmohammadi, Ivan Z. Nenadic, Matthew W. Urban, James F. Greenleaf, Azra Alizad (Physiol. and Biomedical Eng., Mayo Clinic, 200 First St. SW, Rochester, MN 55905, mehrmohammadi.mohammad@mayo.edu), Douglas A. Husmann, Lance A. Mynderse (Urology, Mayo Clinic, Rochester, MN), and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Bladder compliance is defined as bladder's ability to expand in volume without a significant change in pressure. The bladder wall is a multi-layered structure including a muscular layer and fibrous connective tissue. As the percent of connective tissue in the bladder interstitium increases compared to smooth muscle, the bladder becomes more rigid and less capable of expanding during filling. Currently, urodynamic studies (UDS) are considered as the clinical gold standard for bladder compliance assessment. This procedure is invasive and is associated with patient discomfort and is expensive. Ultrasound bladder vibrometry (UBV) is a novel acoustic-radiation-forced-based method for noninvasive assessment of bladder compliance. In UBV, an impulsive acoustic radiation force is focused on the bladder wall (under B-mode ultrasound image guidance) to induce vibrations which may be modeled by as a Lamb wave. High frame rate ultrasound is then utilized to detect the induced waves. This wave motion is then used to estimate the viscoelastic properties and the compliance of the bladder wall. Our results reveal a remarkable agreement between UBV and UDS cystometry measures, suggesting the potential of UBV as a viable clinical tool for the assessment of bladder compliance.

11:45

**2aBA14. Correlation of ultrasound speckle pattern and arrival time errors in shear wave elastography.** Stephen A. McAleavey (Biomedical Eng., Univ. of Rochester, 309 Goergen BME/Optics Bldg., Rochester, NY 14627, stephen.mcaleavey@rochester.edu)

Speckle results from coherent illumination and phase-sensitive detection of echoes from many sub-resolvable scatterers. Speckle provides a texture that allows ultrasonic tracking of tissue motion, which in turn enables the many varieties of ultrasound elastography. A complication is that the presence of speckle can induce a shift in the apparent phase or arrival time of a propagating shear wave. The correlation between shear wave arrival time and the speckle pattern measured with a stationary transmit beam and swept receive beams is demonstrated. Simulated transient shear waves generated in response to acoustic radiation force are tracked with transmit f-numbers of 2 to 4 and receive f-numbers of 2 to 8. The shear wave arrival time error and the lateral first moment of the swept-receive speckle pattern scaled by the shear wave speed are shown to be strongly, though not perfectly, correlated (r ~0.6). These arrival time errors are comparatively insensitive to the amplitude (over a range of $-20$ to 20 $\mu$m) and direction of propagation of the shear wave; e.g., arrival times for $-20$ and $+20$ $\mu$m shear waves differed by $<10\%$ of the total arrival time error. Approaches for the suppression of speckle noise in shear wave elastography are discussed.

12:00

**2aBA15. Concentration of blood components by acoustic radiation force.** Daniel Kennedy (Pharmacology, Western New England Univ., Springfield, MA), Brianna Sporbert, Tyler Gerhardson (Biomedical Eng., Western New England Univ., Springfield, MA), Dane Mealy (FloDesign Sonics, Springfield, MA), Michael Rust (Biomedical Eng., Western New England Univ., Springfield, MA), and Bart Lipkens (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Box S-5024, Springfield, MA 01119, blipkens@wne.edu)

We have recently reported on macro-scale separation of lipids from red blood cells (J. Acoust. Soc. Am. **133**(5, Pt. 2), 3279). In this system, the acoustic radiation force is used to trap red blood cells and lipid particles in a standing wave. Subsequent clumping of red blood cells results in gravitational settling of the red blood cells. Similarly, coalescing of lipids results in rising of the lipids out of solution. We now report on more detailed measurements of the concentration and separation of various blood components, such as red blood cells, white blood cells, and platelets. Porcine blood was used in the experiments and diluted by a factor of ten in phosphate buffered saline. A VetScan HM5 hematology analyzer was used to do the blood count. Inlet flow rate through the device was 16 ml/min, the concentrate flow rate of the blood components was typically about 1 ml/min. The transducer was a 2 MHz PZT-8 operating at 10 W. No lysing was observed in any of the experiments. Results indicate successful capture of red blood cells and white blood cells with separation efficiencies in excess of 90% in a single pass. [Work supported by NSF PFI:BIC 1237723.]

TUESDAY MORNING, 3 DECEMBER 2013 — MASON, 8:30 A.M. TO 10:35 A.M.

## Session 2aEA

## Engineering Acoustics and Structural Acoustics and Vibration: Composite Plates

Andrew J. Hull, Chair
*Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841*

### *Invited Papers*

8:30

**2aEA1. Far-field approximation for a point-excited anisotropic plate.** Elizabeth A. Magliula (NAVSEA Newport, 1176 Howell St., Bldg. 1302, Newport, RI 02841, elizabeth.magliula@navy.mil), J. Gregory McDaniel, and Allan D. Pierce (Dept. of Mech. Eng., Boston Univ., Boston, MA)

An analytic approximation is derived for the far-field response of a generally anisotropic plate subject to a time-harmonic point force acting normal to the plate. This approximation quantifies the directivity of the flexural wave field that propagates away from the force, which is expected to be useful in the design and testing of anisotropic plates. Derivation of the approximation begins with a two-dimensional Fourier transform of the flexural equation of motion. Inversion to the spatial domain is accomplished by contour integration over the radial component of wave number followed by an application of the method of stationary phase to integration over the circumferential component of wave number. The resulting approximation resembles that of an isotropic plate but involves wave numbers, wave amplitudes, and phases that depend on propagation angle. Numerical results for a plate comprised of bonded layers of a graphite-epoxy material illustrate the accuracy of the method compared to a numerical simulation based on discrete Fourier analysis. Three configurations are analyzed in which the relative angles of the layers are varied. In all cases, the agreement is quite good when the distance between force and observation point is greater than a few wavelengths.

8:50

**2aEA2. Optimal design of composite plates based on wave propagation characteristics.** J. Gregory McDaniel (Mech. Eng. Dept., Boston Univ., 110 Cummington Mall, Boston, MA, jgm@bu.edu) and Elizabeth A. Magliula (NAVSEA Newport, Newport, RI)

The present work is concerned with the design of composite plates that are optimized with respect to their wave propagation characteristics. For example, one may wish to design a composite plate such that the attenuation of a dominant wave is maximized. The optimization proposed here considers a composite plate with a specified number of layers. The material properties and thicknesses of the layers are considered as optimization parameters and these parameters are typically constrained in some way. For each choice of the

optimization parameters, complex wave numbers and their associated wave shapes are calculated by using the semi-analytical finite element method developed by others. This method uses a finite element discretization in the thickness coordinate and a propagating wave solution in the lateral coordinates, resulting in a quadratic eigenvalue problem for the complex-valued wave number of each wave that the plate supports. One then computes a scalar cost function based on the complex wave numbers and their associated wave shapes, and proceeds by adjusting the optimization parameters to minimize the cost function. This presentation will describe the computational aspects of the optimization approach and will illustrate its potential by example. [Work supported by ONR under grant N000141210428.]

**9:10**

**2aEA3. Establishing the admissible waves and the steady state response of a rib-stiffened, layered plate structure subjected to high frequency acoustic loading.** Kirubel Teferra and Jeffrey Cipolla (Appl. Sci. Div., Weidlinger Assoc., 375 Hudson St., New York, NY 10014, kirubel.teferra@wai.com)

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. 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. We propose to identify the set of admissible propagating (and attenuating) waves via an eigenvalue analysis as a preprocessing step when executing the pseudo-analytical solution. A significant challenge lies in determining the admissible waves of structures with periodicity in two-dimensions. The Wave Guide Finite Element (WFE) method leads to a two parameter, nonlinear eigenvalue problem, which is extremely difficult to solve. By formulating the problem with the Scale Independent Element (SIE) formulation, the admissible waves can be expressed through a two-parameter quadratic eigenvalue problem. This formulation overcomes the numerical conditioning issues associated with WFE method. This study compares the aforementioned ansatz with that computed by the SIE formulation as well as the improvements in the numerical stability associated with this additional step. The forced response results are compared to a finite element analysis.



**9:30**

**2aEA4. Acoustic scattering from finite bilaminar composite cylindrical shells-three-dimensional solution.** Sabih I. Hayek (Eng Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (NAVSEA Div Newport, Newport, RI)

The acoustic scattering from a finite bilaminar cylindrical shell is analyzed using the exact theory of three-dimensional elasticity. The two lamina are perfectly bonded having the same lateral dimension but have different radii and material properties. The finite shell is submerged in an infinite fluid medium terminated by two semi-infinite rigid cylindrical baffles. The shell has shear-diaphragm supports at the ends $z = 0$, L and is internally filled with an acoustic medium. It is insonified by an incident plane wave at an oblique incidence angle. The scattered acoustic farfield is evaluated for various incident wave wavenumbers, shell thicknesses, shell dimensions, radii, and material properties. A uniform steel shell in water was initially analyzed to study the influence of shell geometries on the scattered acoustic farfield. A second shell made up of an outer elastomer shell bonded to an inner steel shell was also analyzed to study the influence of elastomeric properties on acoustic scattering. [Work supported by NAVSEA Division Newport under ONR Summer Faculty Program.]

## *Contributed Papers*

**9:50**

**2aEA5. Ultrasonic phased array techniques for composite material evaluation.** Hossein Taheri, Fereidoon Delfanian, and Jikai Du (Mech. Eng., South Dakota State Univ., Box 2219 SCEH 216, Brookings, SD 57006, hossein.taheri@sdstate.edu)

The increasing rate of composite materials usage in the industries and researches implies finding the suitable method for testing and evaluation of composite materials. The composites are susceptible to flaws during production and the inspection costs suggest the use of NDT methods. Non-destructive testing is an appropriate method to detect the flaws and anomalies in the materials, however, due to an-isotropic structure of the composite materials, it is required to modify the techniques to find the most appropriate and accurate process for detecting flaws and anomalies in composite materials. The objective of this paper is to evaluate the feasibility and accuracy of nondestructive testing methods with emphasis on ultrasonic phased array technique for the integrity and structural evaluation of composite materials. In this approach three different composite samples were used for testing including one carbon fiber and two glass fiber samples. ultrasound phased array technique evaluated to perform the testing on the composite materials to find the appropriate method and procedure to apply ultrasound phased array techniques on composite materials.

**10:05**

**2aEA6. Elastic response of an orthogonally reinforced plate.** Andrew J. Hull (Autonomous Systems and Technol. Dept., Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil) and Jason M. Maguire (Sensors and Sonar Systems Dept., Naval Undersea Warfare Ctr., Newport, RI)

This paper develops a three-dimensional fully elastic analytical model of a solid plate that has two sets of embedded, equally spaced stiffeners that are orthogonal to each other. The dynamics of the solid plate are based on the Navier-Cauchy equations of motion of an elastic body. This equation is solved with unknown wave propagation coefficients at two locations, one solution for the volume above the stiffeners and the second solution for the volume below the stiffeners. The forces that the stiffeners exert on the solid body are derived using beam and bar equations of motion. Stress and continuity equations are then written at the boundaries, and these include the stiffener forces acting on the solid. A two-dimensional orthogonalization procedure is developed, and this produces an infinite number of double indexed algebraic equations. These are all written together as a global system matrix. This matrix can be truncated and solved resulting in a solution to the wave propagation coefficients, which allows the systems displacements to be determined. The model is verified by comparison to thin plate theory and finite element analysis. An example problem is formulated. Convergence of the series solution is discussed.

10:20

**2aEA7. On fluid-structure interactions of a cloaked submerged spherical shell.** Clyde Scandrett (Appl. Mathematics, Naval Postgrad. School, Spanagel Hall Bldg. 232, Monterey, CA 93943, clscandr@nps.edu) and Ana M. Vieira (Naval Res. Ctr., Lisbon, Portugal)

Backscattering from a cloaked submerged spherical shell is analyzed in the low, mid, and high frequency regimes. Complex poles of the scattered pressure amplitudes using Cauchy residue theory are evaluated in an effort to explain dominant features of the scattered pressure and how they are affected by the introduction of a cloak. The methodology used is similar to that performed by Sammelmann and Hackman in a series of papers written on scattering from an uncloaked spherical shell in the early 1990s. In general, it is found that cloaking has the effect of diminishing the amplitude and shifting tonal backscatter responses. Extreme changes of normal and tangential fluid phase velocities at the fluid-solid interface when cloaking is employed leads to elimination of the "mid-frequency enhancement" near the coincidence frequency for even modestly effective cloaks, while reduction of the "high-frequency enhancement" resulting from the "thickness quasi-resonance" near the cutoff frequency of the symmetric ($S_B^2$) mode require more effective cloaking, but can be practically eliminated by employing a cloak that creates tangential acoustic velocities in excess of the $S_B^2$ mode phase speed near cutoff.

TUESDAY MORNING, 3 DECEMBER 2013 CONTINENTAL 5, 8:00 A.M. TO 11:25 A.M.

## Session 2aED

## Education in Acoustics: Engaging and Effective Teaching Methods in Acoustics

Michelle C. Vigeant, Cochair
*Graduate Prog. in Acoustics, Penn State Univ., State College, PA 16801*

Preston S. Wilson, Cochair
*Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292*

**Chair's Introduction—8:00**

### *Invited Papers*

8:05

**2aED1. Community-based inquiry: An example involving wind turbine noise.** Andrew A. Piacsek and Bruce Palmquist (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

Community-based inquiry (CBI) is an approach to providing an active and collaborative learning environment for students in a science classroom. Inquiry-based learning has been shown to improve critical thinking skills, as well as retention of new concepts [Quitadamo *et al*., Life Sci. Educ. **7**, 327–337 (2008)]. For a typical CBI project, all students in a class work simultaneously on single problem that has significance to the local community. The problem is generally structured as an open-ended investigation that incorporates concepts covered in class. Specific components of the problem can be assigned to groups of students; collaboration among groups helps the class arrive at "big picture" conclusions. At Central Washington University, the introductory calculus-based physics course was recently revised to incorporate a CBI project involving the acoustics of local wind turbines. Wind speed and sound level measurements were taken by faculty at two different sites under different weather conditions. Each student group chose a subset of the data to analyze, collaborating via an Internet-based shared document platform. The design and implementation of this project, as well some assessment results, will be discussed.

8:25

**2aED2. Modifying the just in time teaching strategy for introductory acoustics.** Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd.,Joliet, IL 60431, amorrison@jjc.edu)

Just-in-time-teaching (JiTT) is a teaching and learning strategy where a feedback loop is created between instructor and students in order to focus classroom activities toward concepts the class is having the most difficulty in understanding. The two parts of a JiTT strategy are short exercises done outside of class and the response to the exercises in class. In previous iterations of my introductory course on acoustics, students have been assigned readings to complete and questions to answer based on the assigned readings. Students are required to answer all questions online before the start of class. The instructor is responsible for reading student responses before the start of class and has the opportunity to adjust class activities to address concerns raised by students in their answers. This year, we are modifying the JiTT method to encourage student reflection on the activities completed in class and emphasize critical reading skills. The benefits and challenges of the JiTT method and present modifications are discussed.

**8:45**

**2aED3. Using acoustics to enhance physics education.** Tracianne B. Neilsen and Kent L. Gee (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu)

The authors were recently invited to write a Resource Letter for the *American Journal of Physics* on the role of acoustics in enriching physics education. The Letter contains citations related to textbooks, physical and virtual acoustics demonstrations, software, and instances where acoustical examples can lead to a deeper understanding of, e.g., general wave phenomena. This talk provides a summary of the diverse nature of resources that exist as well as specific tools and examples.

**9:05**

**2aED4. Techniques used to promote active learning in an undergraduate architectural acoustics course.** Michelle C. Vigeant (Graduate Program in Acoust. & Architectural Eng. Dept., The Penn State Univ., Penn State Univ., State College, PA 16801, michelle.vigeant@gmail.com)

The five-year undergraduate architectural engineering (AE) program at Penn State requires all students to take an introductory course in architectural acoustics (AA) and for most students in the program, no further AA courses are required. As a result, it is very important to capture the students' interest early on in the course and use strategies that will improve long-term retention of the material. A number of techniques were used to improve student engagement with the introductory AA course that had an enrollment of 94 students. The first homework assignment was to summarize a recent magazine article in any area of acoustics, which allowed the students to immediately see a real-world application of the subject. During lecture periods, students were encouraged to participate through the use of personal response devices, also known as "clickers." Several different types of clicker questions were used, including review questions from the preceding lecture, questions to prompt participation during the presentation of example problems, and conceptual questions. The results from the multiple choice clicker questions are immediately displayed, which provides useful feedback to both the students and the instructor. Some hands-on activities during the practicum periods were also incorporated to reinforce the content from the lectures.



**9:25**

**2aED5. Experimental learning in acoustics: A project oriented graduate course on architectural acoustics.** Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

The idea of learning-by-doing, known in educational theory as experimental learning, has wide acceptance as an efficient and effective teaching method. The subject of architectural acoustics is particularly well suited for this type of instruction. This talk reports on the current graduate course on architectural acoustics in the graduate acoustics program at The University of Texas at Austin, which attempts to implement an experimental learning approach. The course focuses on in-class demonstrations, site visits, and an extended course project to supplement traditional lectures. The talk will demonstrate how this approach satisfies an experimental learning model which emphasizes a cyclic process of abstract conceptualization (lectures on acoustical theory), experimentation and experience (demonstrations, visits, and projects), and reflective observation (reporting and testing). Specific example course projects are provided that illustrate the breadth and depth of course projects resulting from this approach and perspectives are provided for avenues for improvement using this teaching method.

**9:45**

**2aED6. Development of educational stations for the Acoustical Society of America outreach activities.** Cameron T. Vongsawad, Tracianne B. Neilsen, and Kent L. Gee (Phys. & Astronomy, Brigham Young Univ., 1041 E. Briar Ave., Provo, UT 84604, cvongsawad@byu.net)

One of ASA's outreach activities is hands-on exploration workshops for local school groups and girl scouts troops held during our semiannual conferences. In general, outreach programs have three main purposes: (1) public service, (2) generate enthusiasm and interest, and (3) supplement learning. Despite the good work that has been done in the past, it is apparent that the goals of the current ASA outreach activities could be better achieved through increased efficiency. This project continues the development of the ASA outreach programs held in conjunction with our meetings to better meet the above goals. Specifically, we have developed a structure in which demonstrations are grouped into five-minute stations each pertaining to a different physical system. To increase the ease with which volunteers can assist with the stations, one-page summaries and reference posters outlining the basic principals of each station have been prepared. Volunteers are encouraged to be guided by the interests of the students in their interactive discussions. In addition, we have created a brief introductory presentation for the workshops to explain what can be done in the field of acoustics. These improvements not only provide opportunities to excite student interest but also increase the efficacy of the outreach efforts.

**10:05–10:20 Break**

**10:20**

**2aED7. Computer modeling as teaching tool in underwater acoustics.** Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

One of the challenges of teaching acoustics is to bridge the gap between theory and application. However, laboratory experiments are often too difficult or costly particularly for large-scale phenomena such as ray bending in ocean acoustics. Computer modeling is a low cost, easily accessible way for students to visualize acoustic phenomena without the overhead of laboratory experiments. In the underwater acoustics course at the University of Texas at Austin, students develop an acoustic modeling suite throughout the course culminating in a code to solve the sonar equation for a target in a shallow water environment. Beginning with simple attenuation, students model rays bending due to sound speed profiles, modes, both propagating and leaky, reverberation, target scattering, array steering, and signal processing. Each component builds on previous code and, at the end of the course, the student has a coherent collection of underwater acoustic modeling tools.

## *Contributed Papers*

10:40

**2aED8. Using game-based learning with integrated computer simulation to teach core concepts in underwater acoustics.** Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Game-based learning promises enhanced student engagement coupled with effective pedagogy, which motivated development of WaveQuest, a computer game that employs simulation to assist secondary students in learning core concepts from underwater acoustics. Prior work in physics education found simulations can aid learning by providing environments for inquiry that, through constraints on action imposed by concrete connection to the real world, implicitly scaffold the learning process, yet also support clarifying representations of phenomena unavailable to observers in the real world. The interactive 3D narrative game developed in this work provides motivational context for computer simulations and clarifying representations, which enables and assists students in pursing inquiry-based exploration of the simulated physical environment. At the same time, the level progression and game play naturally scaffold an adaptive learning process. Beyond facilitating this inquiry-based learning process, the game mechanics have been designed to incentivize players toward internalizing the novel concepts and models that are presented, in order to facilitate more rapid acquisition of new cognitive schema. This presentation will describe and demonstrate key elements of the game, with particular attention given to illustrating their relation to this pedagogical strategy. [Work supported by ONR Code 321 US.]

10:55

**2aED9. Development of an online undergraduate course in acoustics at the Berklee College of Music.** Eric L. Reuter (Liberal Arts, Berklee College of Music, 1140 Boylston St., Boston, MA 02215, ereuter@berklee.edu)

Between 125 and 150 students take an introductory course in acoustics each semester at the Berklee College of Music in Boston, MA. As the popularity of acoustics has grown (owing in part to a new minor in acoustics and electronics) limited classroom facilities have made it difficult to meet the demand for sections of the introductory class. Development is nearly complete on a fully online version of the course that will be offered beginning in 2014. This paper will examine the curriculum, development of media assets, and various hurdles in the development process.

11:10

**2aED10. A water-level controlled wind instrument developed by a team-based project for an undergraduate acoustic class.** Seung-Cheol Kang and Dong-Guk Paeng (Ocean System Eng., Jeju National Univ., 102 Jejudaehakno, Jeju 690-756, South Korea, stealyour@nate.com)

During an introductory undergraduate class of acoustics, a team-based project was given to design and implement their own musical instrument, after studying basic concepts of resonance, standing wave, and modes of string, pipe, bar, membrane, and plate. One of the teams designed and implemented "W.L.C WINDY (Water-Level Controlled Wind Instrument)," which the length of the pipe can be adjusted by water level inside the pipe. The lengths satisfies the modes of a stopped pipe (open-closed end) with end correction. There are several advantages of this instrument for education of acoustics; cheap, easy to make, and easy to understand the principles of modes. It costs less than one dollar (a transparency hose, a syringe, an O-ring) to make "W.L.C WINDY". This is how to make the instrument; 1. Connect a transparent hose with a syringe without needle using an O-ring. 2. Fill the hose with colored water for easy visualization. 3. Press the thumb rest of the syringe to make various sounds by controlling the water level. Even though W.L.C WINDY is not optimized to play music, it may contribute to easy demonstration of the principles of the modes for education of acoustics. [Supporting program: NIPA-2013-H0401-13-1007.]

TUESDAY MORNING, 3 DECEMBER 2013 PLAZA A, 8:00 A.M. TO 12:00 NOON

# Session 2aID

# Interdisciplinary: Academic Genealogy (Poster Session)

Steven L. Garrett, Chair
*Grad. Prog. in Acoustics, Penn State, Appl. Res. Lab., P. O. Box 30, State College, PA 16804-0030*

## *Contributed Papers*

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

**2aID1. The Ear Club: Ervin R. Hafter's academic family.** Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), G. Christopher Stecker (Dept. of Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN), Psyche Loui (Dept. of Psych. and Program in Neurosci. and Behavior, Wesleyan Univ., Middletown, CT), and Anastasios Sarampalis (Experimental and Work Psych., Univ. of Groningen, Groningen, Netherlands)

Ervin R. Hafter is a direct academic descendent of Wilhelm Wundt, William James, James Cattell, Robert Woodworth, and Warner Brown. Erv's Ph.D. advisor, Lloyd Jeffress, is one of the most renowned names in binaural hearing theory. For more than 40 years, auditory scientists from around the globe have been traveling to Berkeley, where Erv has been teaching and researching at the University of California, Berkeley, mentoring undergraduate students and Ph.D. students and supervising postdocs. Many of those who worked in Erv's lab have gone on to run labs of their own and have graduated and supervised many of their own students and postdocs, who in turn now run their own labs. This presentation will document the extensive web of connections that all started with the Hafter Lab and the weekly seminar series, the Ear Club, where for decades auditory scientists of all ranges of background and experience have been coming together to talk, listen, and make friends for a lifetime.

**2aID2. Bob Apfel: Acoustics at Yale University and beyond.** Jeffrey A. Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., New York, NY 10038, jketterling@riversideresearch.org), Christy K. Holland (Cardiovascular Inst., Internal Med., Univ. of Cincinnati, Cincinnati, OH), Ronald A. Roy (Dept. of Mech. Eng., Boston Univ., Boston, MA), and E. Carr Everbach (Eng., Swarthmore College, Swarthmore, PA)

Bob Apfel passed away in 2002 after having taught at Yale University for over thirty years. During his tenure at Yale, Bob graduated 28 Ph.D. students, trained 12 post-doctoral students and influenced countless masters and undergraduate students. Bob devoted considerable time and effort to the Acoustical Society of America (his honors include the Gold Medal for Lifetime Achievement, Silver Medal in Physical Acoustics, and the Biennial Award) and his influence is still felt by the continued involvement of many of his students. Prior to his time at Yale University, Bob trained at Harvard University with Frederic V. Hunt and the chain of advisors can be loosely traced back, all at Harvard, to 1802. Although Bob's direct impact on the field of acoustics was prematurely cut short in 2002, his legacy and influence are still growing through the contributions of his students, and the students of his students.

**2aID3. The acoustic genealogy of the Nebraska Acoustics Group at the University of Nebraska—Lincoln.** Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816, lwang4@unl.edu)

The Nebraska Acoustics Group at the University of Nebraska—Lincoln was founded in the year 2000 with the hiring of Dr. Lily Wang into the newly formed Architectural Engineering Program. This poster will review the acoustic genealogy extending to and from this acoustics group, connecting to a number of other acousticians at academic institutions including Princeton University, Georgia Institute of Technology, Pennsylvania State University, Technical University of Denmark, Vassar College, University of Hartford, and Colombia College in Chicago. Connections to architectural acoustic consulting firms will also be highlighted.

**2aID4. Greetings from the West: The people of Veneklasen Associates.** Jerry P. Christoff, John J. LoVerde, and Joe Ortega (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jchristoff@veneklasen.com)

Western Electro-Acoustic Laboratory (WEAL) was founded in 1947 by Paul S. Veneklasen, FASA, for the manufacture of precision acoustical instruments utilizing the Western Electric 640AA Precision Condenser Microphone. When asked in 1951 to provide acoustical consulting services on a jet engine test cell, the architect insisted on dealing with an individual, not a laboratory, and Paul S. Veneklasen & Associates (PSVA) was born. Renamed Veneklasen Associates (VA) after Paul Veneklasen's death in 1996, VA and WEAL have had the privilege of employing excellent associates over 65 years of business. Many within this group have contributed to the science of acoustics, in several cases founding firms to continue the practice of acoustical consulting. Others like the principal author, employed since 1956, have remained at VA continuing to contribute and mentor others in the science of acoustics. All of us have benefited from Paul Veneklasen's approach to consulting projects, his use of corroborating measurements, reporting style and the quest for solutions to acoustical problems. In addition, the firm carries on Paul Veneklasen's tradition of in-house acoustical research to advance the field. The Poster traces this history in terms of its current and former employees and their careers with PSVA and VA.

**2aID5. Katherine Safford Harris' academic family tree.** Fredericka Bell-Berti (Commun. Sci. and Disord., St. John's Univ., 8000 Utopia Parkway, Queens, NY 11439, bellf@stjohns.edu), Lawrence J. Raphael (Commun. Sci. and Disord., Adelphi Univ., Garden City, NY), and Diane Kewley-Port (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

At Haskins Laboratories, Katherine Harris lead a research team of engineers, programmers, technicians, ENT physicians, and fellow scientists to study speech production. As the reputation of this innovative research program grew, students, post-doctoral fellows, and visiting scientists from many universities and international laboratories came to join Kathy in this effort. Her contributions to training doctoral students grew when she became a faculty member at the CUNY Graduate Center. She was a pioneer role model especially—but not exclusively—for women scientists. Her enthusiasm and dedication to science attracted a host of students and young scientists who have gone on to academic research careers in Speech Communication. Kathy nurtured her students, always ready to listen and encourage, guiding but never pushing them through their degrees, and always keeping sight of the person.

**2aID6. One view of the Rudnick-Putterman UCLA-Acoustics dynasties.** Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Appl. Res. Lab., P. O. Box 30, State College, PA 16804, sxg185@psu.edu)

During the second half of the 20th century, the research group headed by Isadore Rudnick, augmented, then extended to this day by Seth Putterman, brought the formalism and techniques of classical acoustics to problems in the quantum mechanics of condensed matter systems. In doing so, Izzy advised 32 Ph.D. students who graduated between 1951 and 1986, half of which went on to academic careers of their own (excluding two of his sons who are also academic scientists), making the UCLA-Acoustics "family tree" extraordinarily dense and complex. Although Putterman's academic roots can be traced all the way back to Stefan, Boltzmann, Ehrenfest, and Uhlenbeck, it appears that Rudnick's inspiration was divine in its origin. Traveling forward from my time as a Rudnick-Putterman graduate student in the mid-1970's, several clusters of my "academic siblings" (fellow graduate students and post-docs) can be identified that exerted a strong influence on acoustics research and education at the Naval Postgraduate School and Penn State and were influenced by the Swift-Wheatley-Migliori (Thermoacoustics) Group at Los Alamos National Laboratory.



**2aID7. Acoustic lineage of the electroacoustic research laboratory at University of Massachusetts Dartmouth.** David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

This poster summaries the academic and industrial lineage of the electroacoustics research laboratory at the ATMC-UMass focusing on the influence and training of mentor Boris Aronov, his use of the energy method for solving transducer problems, and a summary of some of the new generation of students and their research projects. Aronov adapted variational methods and generalized coordinates to solve transducer problems using an energy-balance approach in the 1950's in Russia under the tutorage of Lev Gutin at the Morphyspribor (Marine Physics Research Center), the soviet navy's main Sonar Research Institute located in Leningrad (now Saint Petersburg). Shortly after immigrating to the United States in 1998, Aronov teamed up with David Brown at UMass Dartmouth and BTech Acoustics LLC, building a fruitful partnership involving undergraduate and graduate students focusing on underwater electroacoustic transducer and electroacoustic subsystem problems—ranging from basic research, materials characterization, transducer and subsystem modeling, prototype development, calibration, and manufacturing.

**2aID8. Academic genealogy of Lawrence A. Crum.** Lawrence A. Crum (Appl. Phys. Lab., Univ. of Washington, Ctr. for Industrial and Medical Ultrasound, Seattle, WA 98105, lac@apl.washington.edu)

Lawrence Crum received his Ph.D. in physics from Ohio University in 1967; his advisor was Prof. F. Burt Stumpf. He studied as a postdoc under Prof. F. V. Hunt at Harvard from 1967 to 1968, and held faculty positions in physics at the U.S. Naval Academy from 1968 to 1978, where he supervised Trident Scholars—selected undergraduates who were able to perform the equivalent of a MS thesis. From 1978 to 1992, he held positions in the Department of Physics at the University of Mississippi, where he supervised a number of graduate students toward advanced degrees. Since 1992, he has held positions in the Departments of Electrical Engineering and Bioengineering at the University of Washington, where he has continued to supervise students toward advanced degrees. This poster will attempt to document the many students who have studied under Dr. Crum's tenure during his academic career.

**2aID9. Historical perspective on R. Bruce Lindsay, A. O. Williams, R. T. Beyer, and P. J. Westervelt: Connections and branches surrounding the Brown era.** Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

Starting from a timeline when R(obert) Bruce Lindsay graduated from Brown (1920), there is a lineage of so-called great-grandfather, grandfather, and father advisors leading up to his own research, and branches connecting new Ph.D.'s. His Ph.D. on the atomic models of the alkali metals began in Copenhagen, Denmark, under Niels Bohr and H.A. Kramers in 1922, and was completed at M.I.T. in 1924. Bruce Lindsay returned to Brown University in 1930 (from Yale), became chair in 1934 and Hazard professor in 1936. His interests in atomic physics switched to acoustics during and after WWII, obtained strong support from ONR and kindled a legendary relationship with the ASA, including 20 years as Editor-in Chief, and a gold medal. He brought on board Art Williams (1940), Robert Beyer (1945), and Peter Westervelt (1951). They also received medals. R.T. Beyer's advisor at Cornell, Harry Sack, was a former student of Peter Debye. A.O. William's and P. J. Westervelt's advisors were Bruce Lindsay and Richard Bolt (M.I.T.), respectively. This poster presentation will also include threading together acoustics at Brown in an era including Jordan Markham, Bob Morse, Rohn Truell, Bruce Chick, Wes Nyborg, Charles Elbaum, and Humphrey Maris.

**2aID10. Percy Wilson, acoustical consultant.** Geoffrey L. Wilson (Acoust. (retired), Penn State Univ., 441 West Nittany Ave., State College, PA 16801-4057, glw4@psu.edu)

Percy Wilson was Technical Adviser to The Gramophone magazine in England from 1925 to 1938. He wrote a monthly column called Technical Talk and also wrote articles for the Wireless Magazine. He was very well known in the UK Audio industry. He was a senior civil servant at the Board of Education and in 1937 was appointed Assistant Secretary of the Ministry of Transport, in administrative charge of the trunk roads division. Because of pressures of work due to the probable imminence of war with Germany, he was forced to give up most of his outside activities, but was able to resume them after the end of WW2 and his subsequent retirement from the Civil Service. He was Technical Editor of The Gramophone for many more years, and was active in the Audio Engineering Society, being one of the founders of the British Section. As the self-appointed family historian I will endeavor to present highlights of his life from the time of his birth in Halifax, Yorkshire on March 3, 1893, to his demise in Oxford on April 30, 1977, focusing primarily on his technical work.

**2aID11. The academic family tree of Ingo Titze.** Brad Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., Rm. 214, Tucson, AZ 85721, bstory@u.arizona.edu), Eric Hunter (National Ctr. for Voice and Speech, Salt Lake City, UT), and Ronald Scherer (Dept. of Commun. Sci. and Disord., Bowling Green State Univ., Bowling Green, OH)

Pursuing a deep interest in understanding the complexities of human sound production, particularly singing, Ingo Titze received his Ph.D. in Physics under the tutelage of Bill Strong at Brigham Young University in 1972. His dissertation on computational modeling of vocal fold vibration launched a career in research and teaching that has had a profound influence on the field of voice and speech science. Although his research alone is prodigious, Ingo Titze created an academic family that stretches far and wide. This presentation will trace the careers and research of the many students and scientists that were trained by Ingo Titze or influenced by his training.

**2aID12. Genealogy of the National Center for Physical Acoustics.** Josh R. Gladden, James Sabatier, and Henry E. Bass (Phys. & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu)

NCPA traces its roots to the Physical Acoustics Research Group which grew from the Department of Physics and Astronomy. The current Department of Physics and Astronomy acoustics effort was initiated with Prof. F. Douglas Shields who received his Ph.D. at Vanderbilt University under the direction of Prof. Robert Lageman. Professor Shields came to the University of Mississippi in 1959. He directed the dissertation of 10 Ph.D. students and seven Masters students. In 1964, Prof. Shields was joined in the department by Prof. Roy Arnold who studied the ultrasonic properties of solids. Professor Arnold directed the dissertation of two Ph.D. students and six Masters students. In 1970, Prof. Randall Peters and Prof. Henry Bass joined the faculty; Bass doing acoustics in gases with a Ph.D. from Oklahoma State under Tom Winter and Peters doing solid state acoustics. The following year, Ron Carter joined the group doing solid state acoustics. Before his passing in 2008, Professor Bass directed the dissertation of nineteen Ph.D. students and eight Masters students. Professor Larry Crum, a student of Prof. Stump at Ohio State, joined the faculty in 1978. In 1986, an act of Congress established the National Center for Physical Acoustics and over a period of two years, PARGUM was transferred to NCPA. Professor Ralph Goodman joined NCPA as Director in 1989. A listing of all additional faculty and scientists to join NCPA and the students directed by all will be updated to 2013 and presented.

**2aID13. Genealogy of Kenneth N. Stevens and the Speech Communication Group at the Massachusetts Institute of Technology.** Helen Hanson (ECE Dept., Union College, 807 Union St., Schenectady, NY 12308, helen.hanson@alum.mit.edu) and Stefanie Shattuck-Hufnagel (Speech Commun. Group, MIT Res. Lab. of Electronics, Cambridge, MA)

Ken often told a story to his advisees, illustrating how a seemingly insignificant event could change the course of a life. When Ken began his graduate studies at MIT, Leo Beranek learned that he had taken an acoustics class as an undergraduate and recruited him as a teaching assistant. Beranek was doing research on speech acoustics, and soon Ken was one of his research assistants. Thus, Ken's decision to take an undergraduate acoustics course led him to his lifetime of work in speech communication, in which he found boundless pleasure. After completing his Sc.D. in 1952, Ken joined the MIT Faculty. His first doctoral student was James Flanagan. He supervised at least 31 more doctoral students and scores of post-doctoral, masters, undergraduate, and visiting students over 53 years. Ken collaborated with MIT colleagues (e.g., Morris Halle and Jay Keyser, Linguistics Dept.) and many researchers well beyond 77 Massachusetts Ave (e.g., Gunnar Fant, KTH, Sweden). His interest in how the physics of the vocal tract interacts with contrastive phonological categories unequivocally changed the field of Speech Communication, as well as the lives of his many students and collaborators who have gone on to establish their own laboratories and research programs.

**2aID14. Roots and branches of the acoustics program at Brigham Young University.** Derek C. Thomas, Kent L. Gee, Tracianne B. Neilsen, Timothy W. Leishman, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N223 ESC, Provo, UT 84602, derekcthomas@gmail.com), Jonathan D. Blotter, Scott L. Thomson (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT), and William J. Strong (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Acoustics research at Brigham Young University has a rich history extending back over half a century. The tradition of acoustics continues to this day as BYU is home to one of the largest student chapters of the ASA. The academic roots of current and former faculty will be presented. The locations of the branches (students) will also be presented to illustrate the growth and development of the program and the impact on the broader acoustics community.

**2aID15. One descendant of David Blackstock of the University of Texas at Austin.** Bart Lipkens (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Box S-5024, Springfield, MA 01119, blipkens@wne.edu)

In the first academic genealogy session, held at the 153rd meeting of the Acoustical Society of America in Salt Lake City in 2007, David T. Blackstock displayed a poster detailing the Harvard academic tree leading back to Helmholtz [J. Acoust. Soc. Am. **121**(5, Pt. 2), (2007)]. David also included the tree going up and discussed his Ph.D. and M.S. students. The author of this poster, Bart Lipkens, was one of David's Ph.D. students. I was a faculty at Virginia Commonwealth University for two years, where I had one Ph.D. student, Shaozeng Dong. I then became a faculty at a predominantly undergraduate teaching institution, Western New England University in Springfield, Massachusetts. Western New England offers M.S. programs but limited Ph.D. opportunities. Even though my efforts are primarily focused on undergraduate teaching, I continue an acoustics research program, built mostly around M.S. and undergraduate students through a very active program of REUs (research experiences for undergraduates). This program has led several students to pursue further graduate education and careers in acoustics. A review of these students will be given.

## Session 2aMU

## Musical Acoustics and Education in Acoustics: Experiments and Laboratory Curricula for Undergraduate Courses in Musical Acoustics

Jack Dostal, Chair
*Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109*



### *Invited Papers*

**8:00**

**2aMU1. Making music: A serious attempt at designing new instruments by undergraduate students.** Thomas R. Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu), Joshua Almond (Dept. of Art and Art History, Rollins College, Winter Park, FL), and Daniel Crozier (Dept. of Music, Rollins College, Winter Park, FL)

We describe a course in musical acoustics that required undergraduate students to design and build unique musical instruments, compose music for ensembles of them, and then perform the compositions in a public concert. Unlike most courses in musical acoustics which require the students to build home-made instruments as a final project, the construction of the instrument and composing original music were the primary goals of the course. The instruments were required to be artistic, visually interesting and play a pitch collection from the Western scale. We will describe the challenges and successes, show examples of the instruments, and review the lessons learned.

**8:20**

**2aMU2. Flute measurements in a physics of music lab.** Randy Worland (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Physics of Music students often benefit from laboratory exercises that make use of real musical instruments, in addition to the more traditional labs that are designed to illustrate physical principles as simply as possible. These “real instrument” labs help bridge the gap between idealized cases and the musical instruments the students are familiar with. Modern woodwinds are particularly challenging in this regard due to the complex set of keys, levers, and mechanical linkages that tend to obscure the underlying acoustics of these instruments. Among the woodwinds, the flute is relatively simple, both mechanically and acoustically, and thus provides an excellent subject for a real world woodwind study. Laboratory exercises are described in which the flute’s tone hole locations and diameters are measured. The data are analyzed in terms of the acoustics of open cylindrical tubes, revealing the logical order behind the spacing and use of the holes to play the lowest chromatic octave, as well as the higher registers of the flute. The measurement techniques and analysis are presented along with the pedagogical role of the experiment.

**8:40**

**2aMU3. An interactive audio synthesizer for investigating formants and timbre.** Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

Computer-based investigations of the nature of formants and their role in characterizing vowel sounds and the timbre of musical instruments will be described. One version of this activity requires students to obtain detailed spectra of different vowel sounds, measure relative amplitudes of harmonics in the recorded sounds, and perform additive synthesis using Mathematica to recreate the different vowels. Students can investigate the minimum number of harmonics and formants needed to distinguish vowels. A new version of this activity relies on a recently developed interactive audio synthesis program that reduces the emphasis on tedious measurements and the need to edit Mathematica notebooks, allowing students to devote intellectual effort on investigation. The new program and the associated lab activity will be tested on students during Fall quarter of 2013 as part of the “Physics of Musical Sound” course at Central Washington University.

### *Contributed Paper*

**9:00**

**2aMU4. A carillon bell laboratory in an introductory physics of music class.** Jack Dostal (Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

The Physics of Music is a lecture and laboratory class which satisfies a quantitative physical science requirement. It is open to students of all majors. Students taking the class may or may not have musical backgrounds, and generally are not majors in the physical sciences. In this laboratory exercise, students record sounds of the bells of the Harris Carillon in the tower of WFU’s Wait Chapel. Using Audacity and other software, students generate frequency spectra of different bells. The prominent presence of minor thirds in the spectrum makes for a useful comparison between spectra of bells and typical concert instruments. In this lab, students identify spectral features of individual bells, find similarities and differences between different bell spectra, and observe the decay of spectral features over time. In this talk, I will describe some of the activities performed by the students in the lab and comment on some of the challenges of performing this laboratory exercise with students of varied musical and scientific backgrounds.

## *Invited Paper*

**9:15**

**2aMU5. A laboratory investigation of the effect of trumpet mouthpieces in an introductory musical acoustics course.** Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd.,Joliet, IL 60431, amorrison@jjc.edu)

In our introductory musical acoustics course for non-science majors, a major goal of the course is to introduce students to the principles of scientific thinking and investigation. Many students have preconceived ideas or beliefs about factors influencing the sounds produced by musical instruments. It is the role of the instructor to help students frame their ideas into scientifically testable experiments, which can be conducted in the laboratory. In our course, we have designed a laboratory activity to challenge students to explore the role of the mouthpiece of brass musical instruments. A parametric model of a trumpet mouthpiece is customized by students in order to investigate the effects of changing mouthpiece geometry on the trumpet acoustics. The models are manufactured on a 3D printer and used in the laboratory to make input impedance measurements. Students compare their quantitative results to the qualitative conceptions they held before completing the laboratory investigation.

TUESDAY MORNING, 3 DECEMBER 2013 CONTINENTAL 9, 8:10 A.M. TO 12:00 NOON

# Session 2aNS

# Noise and ASA Committee on Standards: Standardization in Soundscape

Brigitte Schulte-Fortkamp, Cochair
*Technische Universität Berlin, Einsteinufer 25, Berlin 10178, Germany*

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

**Chair's Introduction—8:10**

## *Invited Papers*

**8:15**

**2aNS1. The urgent need for standardization of soundscape.** Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de), Klaus Genuit, and André Fiebig (Tech. Management, HEAD Acoust. GmbH, Herzogenrath, Germany)

In recent years, the soundscape approach gains increasingly in importance. It has shown its potential dealing with environmental noise issues, which conventional noise control approaches frequently fail to cope with. The soundscape concept is applied to comprehend the way humans experience and understand their acoustic environments. It does not simply reduce noise to an averaged acoustic quantity evoking unpleasantness estimated by statistical probabilities, but it considers noise as a valuable resource, which can be purposefully utilized. The standardization of "soundscape" in ISO 12913 is currently in progress. The standard will include definitions, terms and a conceptual framework illustrating that soundscape goes beyond physics and psychoacoustics, since it reflects human perception including cognitive aspects, context and interaction. The paper describes the progress in standardizing soundscape and the implications of the standard with its potential relevance to noise policy. Moreover, it discusses how to meet the challenge to standardize and harmonize the soundscape approach without losing the openness and multidimensionality of the soundscape idea.

**8:35**

**2aNS2. Types of soundwalks and their applications in soundscape design.** Gary W. Siebein (Architecture, Univ. of Florida, 625 NW 60th St. Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Keely M. Siebein, and Gary Siebein (Siebein Associates, Inc., Gainesville, FL)

Seven types of soundwalks are described with examples of the data obtained and the application of each to a case study project. The first type is one where an observer walks through an environment and just listens to get an introduction to the environment or an inspiration for later work. The second type is one where the observer walks with a group of stakeholders to listen to or elicit observations about the environment. The third type of soundwalk is one in which qualitative data in the form of questionnaire responses or quantitative data in the form of sound level measurements or recordings are made. The fourth type is an undercover soundwalk where observations or

data are recorded in a way that the observer is not identified as one is not a normal participant in the soundscape. The fifth type is one where stakeholders experience an existing environment that has elements being considered for use in their project. A sixth type of soundwalk is one that occurs in a computer simulation where one experiences a virtual environment. The seventh type is one that combines two or more of the individual types of soundwalks in a composite soundwalk.

**8:55**

**2aNS3. Evaluation of indoor and outdoor soundscapes—The benefit of combining soundwalks and laboratory tests.** Kay S. Voigt (Inst. of Fluid Mech. and Eng. Acoust., Technische Universität Berlin, Einsteinufer 25, Berlin 10587, Germany, kay.s.voigt@gmx.de)

Experiences and expectations of listeners have a significant influence to ratings of soundscapes. Sounds do not only occur on a single level of perception to peoples' ears, they appear with a setting of information of the source(s) to the "listening background" of the individual at a certain location. Although these possible combinations seem to be infinite, common judgments are unfolded on appropriateness of sounds on places. For the evaluation of two public indoor and two public outdoor locations in the city of Berlin, the soundwalk has been used with students in a series of five consecutive years to investigate predominant aspects of rating soundscapes. This tool provides a substantial data acquisition of the environment with technical measurements and documentations as well as ratings and appraisals by the participants. The data-triangulation enables to detect contextual moderators of the participants' evaluation by adjusting situational focus and importance. In subsequent laboratory tests—with the recorded stimuli—this connotation of non-acoustical features is discovered, especially by their absence, also the underlying attitude towards sound sources is identified. Finally, a reflecting discourse with the attendees on their ratings shows the need of elucidation of the meaning of individual use of words and phrases.



**9:15**

**2aNS4. Combining elements of the soundscape—Lessons from the recording studio.** Alex Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

So many soundscapes—found, archived, preserved, or composed—consist of multiple sound elements. The interaction among these individual sounds can conflict and obscure, or complement and harmonize. The creation of sound recordings through the multitrack production process offers a point of view into how an overall soundscape can be successfully created, and its multiple sounds effectively orchestrated.

**9:35–9:45 Break**

**9:45**

**2aNS5. Structural equation modeling of soundscape perception based on urban contexts.** Jin Yong Jeon and Joo Young Hong (Dept. of Architectural Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul 133791, South Korea, jyjeon@hanyang.ac.kr)

The aim of this study is to explore influential factors on urban soundscape perception. Objective and subjective assessments of the soundscapes in various urban spaces were conducted to characterize soundscape contexts using soundwalks. Evaluations were conducted at 21 locations, including residential, office, park, and commercial spaces in Seoul: the participants assessed their perceptions of both sound and landscape environments. In addition, laboratory experiments using audio-visual stimuli were carried out to validate the dominant soundscape factors derived from the soundwalks. From the results, a perception model of urban soundscapes consisting of physical and subjective indicators was suggested using structural equation modeling.

**10:05**

**2aNS6. Standards vs off-standards in the soundscape design process.** Juergen Bauer (Dept. of Architecture, Waterford Inst. of Technol., Granary, Hanover St., Waterford, Co Waterford 00000, Ireland, jbauer@wit.ie)

Standards are widely acknowledged agreements that make issues and procedures comparable, compatible, and safe. Standards, therefore, help speak a common language and are considered a means of quality assurance, particularly in the field of design. However, in respect to Soundscape research being regarded as a subject of acoustic ecology, there is a limitation: How can standardization assess and evaluate Soundscape entities in an appropriate way which takes ecologic diversity into account? Can descriptors be found which indicate Soundscape qualities other than by measured sound pressure levels, or by surveyed human perception of pleasantness and annoyance? While it is fully acknowledged that these indicators will continue to play a key role in the discussion on Soundscape quality, they fail to address the distinction and the specific "atmosphere" of a specific Soundscape entity. "Atmosphere," as suggested by Gernot Boehme (1993), is "the inbetween" of environmental qualities and human states. By this definition, a specific "atmosphere" is bound to a specific space. This paper explores the potential of using descriptors of "atmosphere," such as in the architectural and urban design process, to help further expand the specification of Soundscape standards, but also to exploit the concept of off-standard Soundscape solutions.

**10:25**

**2aNS7. Psychoacoustics triggering the soundscape standardization.** Klaus Genuit (Tech. Management, HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, klaus.genuit@head-acoustics.de), Brigitte Schulte-Fortkamp (Inst. Tech. Acoust., Tech. Univ. Berlin, Berlin, Germany), and André Fiebig (Tech. Management, HEAD Acoust. GmbH, Herzogenrath, Germany)

Before the start of COST Action TD0804 (network on soundscape of European cities and landscapes) in 2009, the Working Group 54 of ISO/TC 43/SC 1 was established to start consideration of a standardized protocol for assessment of soundscape. The progress is slow since many different protocols are available, which make it difficult to define a harmonized protocol reflecting diverse requirements. No protocol covers all needs, which is characteristic in research dealing with quality of life, where space, social and built environment and culture are often very different. Unfortunately, there is still a lack of open-mindedness to use new soundscape techniques, e.g., in noise action plans and for the protection of quiet areas. The situation differs, however, broadly from country to country. The

Working Group 54 of ISO/TC 43/SC 1 works on reaching consensus to provide the necessary stimulus for further worldwide progress. Currently, the ISO WG 54 is working on measurement procedures and techniques that will trigger the procedures in soundscape research. Psychoacoustics and explorative interviews play a major role, and the triangulation of the respective data is offering consolidated findings. This paper will present related techniques and will discuss their advantages.

## *Contributed Papers*

**10:45**

**2aNS8. Educating for soundscape design.** Fernando J. Elizondo - Garza (Acoust. Lab., FIME-UANL, Calle 8 #422, Col. Villazul, San Nicolas, N.L. 66420, Mexico, fjelizon@hotmail.com)

The development of standardization on soundscapes leads to produce the need of experts and designers of acoustic spaces that be educated in a wide cultural and technological base. Although there is nothing new for well educated acousticians, we cannot forget that people involved in architectural or noise control problems are not always acousticians, thus they do not have a full scope of the possible solutions neither of the listeners expectations about sound environment. Engineering, technology, economics, society, diversity, education, psychology, communication, interdisciplinarity, workgroups, ethics, and other educational issues necessary for soundscapes design are discussed in this paper. Many of these aspects are difficult to be included in the curricula directly, so they must be handled as homework and extracurricular activities, under the direction of an acoustics professor interested in the generation of better sound spaces for the welfare of society.

**11:00**

**2aNS9. Recent developments in an old problem: Helping communities deal with noise.** Alice H. Suter (Suter & Associates, 1106 NE Tillamook St., Portland, OR 97212, ahsuter@comcast.net) and Monica S. Hammer (Network for Public Health Law, Univ. of Michigan School of Public Health, Ann Arbor, MI)

This presentation will suggest areas where professionals in acoustics can collaborate with policy makers to remediate their noise problems and promote an environment more conducive to the public health and welfare. Research over recent years has confirmed and elucidated the adverse effects of noise on health in ways that earlier cross-sectional studies had indicated, but with the added power of a prospective approach and extensive controls. The implications of these findings should be considered widespread and serious. Lacking a coordinated federal program, U.S. Cities and counties are attempting to grapple with their noise issues, without federal support and often with limited technical expertise. Despite passage of the Noise Control Act of 1972 (subsequently de-funded), millions of Americans remain exposed to levels of environmental noise that are harmful to their health and welfare.

**11:15**

**2aNS10. Delivering noise information to communities in presentations and with asynchronous media.** David Dubbink (Noise Management Inst., 864 Osos St., San Luis Obispo, CA 93401, dubbink@noisemanagement.com)

Working with communities to resolve noise problems demands not just accurate data about noise impacts but also effective means of exchanging information. The Interactive Sound Information System (ISIS), developed by the author, represents one means of presenting noise impacts data. Central to the ISIS concept is the idea that actual noise examples, shaped to reflect local situations, are the very best way to increase community understanding of noise management issues. Such a strategy works well in a public hearing or community workshop setting. Presentations can be structured to address local concerns and recorded sounds can be accurately calibrated. However, alternate modes of communicating information have been developed that are asynchronous and individual. The question is how to adapt presentations to such delivery systems and retain the reproducibility and accuracy of written reports or professionally moderated presentations. The paper describes current mechanisms for transmitting noise management data and explores the issues involved in adapting delivery to hand-held devices or other modes of individual access.

**11:30–12:00 Panel Discussion**

## Session 2aPAa

# Physical Acoustics and Structural Acoustics and Vibration: Acoustics of Pile Driving: Measurements, Models, and Mitigation I

Kevin M. Lee, Cochair
*Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

Karl-Heinz Elmer, Cochair
*OffNoise-Solutions GmbH, Leinstr. 36, Neustadt a. Rbge. 31535, Germany*



**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**2aPAa1. Offshore pile driving noise—Prediction through comprehensive model development.** Marcel Ruhnau, Tristan Lippert, Kristof Heitmann, Stephan Lippert, and Otto von Estorff (Inst. of Modelling and Computation, Hamburg Univ. of Technol., Denickestraße 17, Hamburg, Hamburg 21073, Germany, marcel.ruhnau@tuhh.de)

Offshore wind energy is one of the most potent among renewables and thus the worldwide number of offshore wind turbines increases rapidly. The foundations of the wind turbines are typically fastened to the seabed by impact pile driving, which comes along with a significant amount of waterborne noise. To protect the marine biosphere, the use of noise mitigation systems, like bubble curtains or cofferdams, may become necessary. In this context, the model-based prediction of underwater sound pressure levels as well as the design and optimization of effective sound mitigation measures by using numerical models is one of today's challenges. The current work presents a modeling approach that consists of a near field finite element model and a far field propagation model. Furthermore, it has been found necessary to generate a benchmark to allow for a qualitative and quantitative comparison between the manifold modeling approaches that are currently developed at various institutes and companies.

**8:25**

**2aPAa2. Model for underwater noise radiated by submerged piles.** Todd Hay, Yurii A. Ilinskii, Kevin M. Lee, Evgenia A. Zabolotskaya, and Mark F. Hamilton (ARL:UT, Appl. Res. Labs., P.O. Box 8029, Austin, TX 78713-8029, hayta@arlut.utexas.edu)

There is concern that underwater noise generated by marine construction activities and radiated by towers supporting offshore wind turbines may disturb marine mammals, or interfere with passive sensors and communication equipment. In order to understand these effects a semi-analytic frequency-domain model was developed previously for the sound radiated in the water column by a pulsating cylindrical structure embedded in horizontally stratified layers of viscoelastic sediment. This model was in turn coupled to a parabolic equation code for long-range propagation over range-dependent environments [Hay *et al*., J. Acoust. Soc. Am. **133**, 3396 (2013)]. A time-domain version of this model is now presented which enables simulation of impulsive sound sources such as those due to underwater pile driving, and pulsed tonal sources appropriate for use in a finite-sized laboratory tank. In order to validate the model a scaled physical model, consisting of a laboratory tank and metallic cylindrical tube driven in the high kilohertz frequency range, was constructed. Simulations will be presented for a variety of sound sources, and preliminary comparisons with measurements from the scaled model experiments will be made.

**8:45**

**2aPAa3. Effect of the transient fluid-structure interaction on sound radiation from a partially submerged cylindrical pile.** Mardi C. Hastings (Georgia Inst. of Technol., George W. Woodruff School of Mech. Eng., 801 Ferst Dr., Atlanta, GA 30332-0405, mardi.hastings@gatech.edu)

A time-domain structural acoustics model of a partially submerged cylindrical pile excited by an impact force has been developed to predict underwater sound radiation for arbitrary field conditions. Radial boundary conditions are formulated using the velocity potential so pressure waveforms near the pile wall can be estimated; however, the rapid decay observed in measured waveforms is not easily predicted because radiation loading on the structure is also transient. The sudden acceleration of the wall initially results in a net flow of energy from the structure into surrounding fluids. Subsequently part of this energy radiates to the far field as a compressional wave; however, as the forced wall motion decreases and the spatial pressure gradient at the wall reverses direction, fluid accelerates back to the pile and re-excites the wall. A model for this transient fluid-structure interaction was formulated and integrated into the time-domain structural acoustics numerical model. By accounting for energy flow back into the structure, predicted pressure waveforms are in good agreement with field data. Results indicate that a significant portion of the kinetic energy remains in the near field and does not radiate sound, and that temporal characteristics of the impact force may influence far field sound radiation.

9:05

**2aPAa4. Numerical simulations of hydrosound and ground vibrations during offshore pile driving.** Christian Kuhn (Technische Universität Braunschweig, Inst. for Soil Mech. and Foundation Eng., Beethovenstr. 51 b, Braunschweig 38106, Germany, c.kuhn@tu-braunschweig.de)

Most foundations for offshore wind turbines are currently constructed with monopiles. During the installation, the monopile is driven into the ground using a hydraulic hammer. A pressure wave is induced into the pile and passes through the pile shaft which as a result expands radially and induces underwater noise in the surrounding water. Similarly, a shock wave is introduced into the ground which propagates there spherical. In a numerical simulation the driving process is simulated. The focus is on the energy radiated from the pile into the surrounding water-/ground continuum. As part of a research project, an offshore test was performed. During this, the underwater noise in the water as well as the soil vibrations on the seabed were measured. The numerical results are compared with these measurement results.

9:25

**2aPAa5. A model for underwater sound levels generated by marine impact pile driving.** Alexander O. MacGillivray (JASCO Appl. Sci., 2305–4464 Markham St., Victoria, BC V8Z7X8, Canada, alex@jasco.com)

Marine impact pile driving generates very high underwater sound pressures that can be harmful to aquatic life. Environmental assessments for pile driving projects typically require advance estimates of acoustic impact zones for marine mammals and fish. A computer model has been developed to predict the radiated acoustic field from impact driving of cylindrical piles. The stress wave in the pile is predicted using a force-generator model of the hammer-pile system. The force-generator model is coupled to a 1-D finite-difference model of longitudinal stress waves in a cylindrical pile. The radiated pressure is computed by matching the velocity boundary condition at the pile wall using a superposition of monopole sources distributed over the length of the pile in a layered 2-D fluid medium. The transfer function for the monopoles is computed using the near-field Hankel transform for radial particle velocity at the pile wall. Standard ocean acoustic modeling techniques are used to compute the Mach wave propagating away from the pile. As an example, predictions of the model are compared to field measurements obtained in a riverine environment.

9:45

**2aPAa6. Prediction of offshore impact pile driving noise using numerical and analytical approach.** Huikwan Kim, James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, hkkim524@my.uri.edu)

Noise generated by offshore impact pile driving can radiate into and propagate through the air, water, and sediment. Predicting noise levels around the support structures at sea is required to estimate the effects of the noise on marine life. We have been investigated acoustic impact using our previously verified coupled Finite Element (Commercial FE code ABAQUS) and Monterey Miami Parabolic Equation (2D MMPE) models [J. Acoust. Soc. Am. **131**(4), 3392 (2012) and J. Acoust. Soc. Am. **133**, 3419 (2013)]. A simple analytical model for the impact pile driving noise is developed to reduce computation cost and modeling effort when compared to the numerical model. In the present study, we are developing a MATLAB based analytical model to calculate the Receive Levels (RL) in water. This analytical model solves simplified Donnell's equation, i.e., governing equation for the thin cylindrical shell using the modal superposition approach assuming azimuthal symmetry. It solves the Helmholtz or Kirchhoff integral equation to evaluate the acoustic pressure field. We will compare the results of the coupled FE-MMPE numerical model and the MATLAB based analytical model and discuss their advantages and disadvantages. [Work sponsored by the Link Foundation Ocean Engineering and Instrumentation Ph.D. Fellowship program.]

10:05–10:20 Break

10:20

**2aPAa7. Toward a standard for measurement of underwater noise from marine impact piling.** Stephen Robinson (National Physical Lab., Hampton Rd., Teddington TW110LW, United Kingdom, pdt@npl.co.uk), Christ de Jong (Acoust. and Sonar, TNO, The Hague, Netherlands), Andreas Mueller (Müller-BBM GmbH, Hamburg, Germany), and Pete Theobald (National Physical Lab., Teddington, United Kingdom)

Marine impact piling is a significant source of low-frequency impulsive noise in the ocean and is typically used to position piled foundations in relatively shallow water for offshore construction, for example, offshore wind farms, bridge supports, etc. However, there are currently no international standards to define appropriate measurement methodologies, and there is a need to underpin incipient regulation with appropriate measurement standards. In the scientific literature, attempts to report the measured noise levels can be difficult to compare because different metrics are often used (peak-to-peak pressure, rms pressure, sound exposure level, etc). Furthermore, simple assumptions about equivalent point sources are often used in measurements and modeling without sufficient validation. Agreed acoustic metrics and a common way of deriving a meaningful measure of source output are urgently required. This paper describes work to address this need within the International Standards Organisation, specifically Working Group 3 of Technical Committee 43, Sub-Committee 3 (ISO TC43 SC3 WG3). Work to develop a new standard builds upon the expertise already gained by researchers in a number of countries (for example, United Kingdom, Netherlands, and Germany). A description is given of the methodology currently proposed, and the process for engagement with experts within different countries.

10:40

**2aPAa8. Results from background noise measurements in the North Sea.** Max Schuster (DW-ShipConsult, Lise-Meitner-Str. 1-7, Schwentinental 24223, Germany, schuster@dw-sc.de)

In Germany, 12.8 GW offshore wind power shall be installed until 2023, which involves that at minimum 200 wind turbines need to be erected each year. As the majority of the turbines' foundations are based pile driving vast amounts of high level impulsive noise are introduced into the North- and Baltic Sea. Until today, research on pile driving noise largely focuses on sound propagation and noise mitigation in close vicinity of the pile. Further effects of impulsive noise in larger distances over 10 km are rather unknown due to a lack of knowledge on the propagation and on the effects of noise at levels below injury. The German Agency for Nature Conservation has founded a campaign to measure underwater noise in the marine protected areas of the German EEZ. In this course, 10 noise recorders are simultaneously deployed in the large protected area Sylt Outer Reef. At least three wind farms are to be installed in the area's vicinity; therefore, recordings of background noise during the installation phase allow a systematic investigation of impulsive noise propagation over large distances. Results in time domain and in frequency domain are compared with results from commonly available calculation codes.

**11:00**

**2aPAa9. Properties influencing the transmission loss and mitigation of the underwater sound from marine pile driving.** Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng. Dept., Univ. of Washington, Seattle, WA 98105, dahl@apl.washington.edu)

The transmission loss of underwater sound generated by impact or vibratory pile driving, combined with the potential mitigation of this sound, ultimately determine the regulatory zone-of-impact on marine life. For impact pile driving the pile source assumes properties of a "phased array" as shown in Reinhall and Dahl [J. Acoust. Soc. Am. **130**, 1209–1216 (2011)], where sound radiation arises from the state of radial expansion which travels down the pile at a speed which is supersonic relative to the speed of sound in water. The acoustic field that develops is shown to be the dominant contributor to peak underwater sound pressure. An important range scale can be identified that is useful for both guiding measurements, and for transmission loss modeling. For vibratory pile driving, measurements made at close range (within about a one water depth) show considerable low-frequency content that can be below the cut-off frequency of the waveguide. Some key properties of impact and vibratory pile driving will be discussed that govern the nature of transmission loss from, and influence mitigation strategies for, these sound sources. These properties are based on measurements made at marine construction sites in Puget Sound, Washington.



## *Contributed Papers*

**11:20**

**2aPAa10. Transmission loss for vibratory pile driving in shallow water: Modeling and field measurements for a Puget Sound location.** Dara M. Farrell and Peter H. Dahl (Appl. Phys. Lab., Dept. of Mech. Eng., Univ. of Washington, Henderson Hall, Seattle, WA 98105, daraf@uw.edu)

Measurements of the underwater noise from vibratory pile driving were collected at a marine construction site in January 2013, during which 0.76 m steel piles were driven in shallow water (less than 10 m). The sound field was simultaneously measured at three locations at distances 200 m (1 hydrophone) and 400 m (2 hydrophones) from the pile driving location and roughly 150 m to 300 m from the shore. In the Practical Spreading Model (PSM), transmission loss is 15 log10(R/R0): R is the range to which TL is calculated, and R0 is the range for a close range measurement, typically O(10 m) from the source. Models such as the PSM that do not account for bathymetric or sediment properties could not be expected to predict the anomalously high transmission loss (TL) that was observed between the 200 m location and one of the 400 m locations. Important bathymetric features (of order 10 m in range, 1 m in depth) and sedimentary properties for the area were incorporated into a Parabolic Equation model; modeled TL was compared to observed TL for third octave bands. Using a Geographic Information System tool, modeled TL was visualized for the area. [Work supported by Washington Sea Grant.]

**11:35**

**2aPAa11. Depth dependence of the intensity vector from impact pile driving.** David R. Dall'Osto and Peter H. Dahl (Mech. Eng., UW-Seattle, 914 N 38th St., Seattle, WA 98103, dallosto@u.washington.edu)

The direction of the time-integrated acoustic intensity vector defines active intensity streamlines which are perpendicular to this vector. For impact pile driving, these intensity streamlines depend on both depth and time. The time-integrated intensity vector (active intensity) has been shown to change direction over the initial downward traveling Mach cone, the upward traveling reflection of this from the bottom of the pile, and subsequent downward traveling reflection from the top of the pile, referred to as phases in Dahl and Reinhall [J. Acoust. Soc. Am. **134**, EL1 (2013)]. In this study, vertical line array data are used to demonstrate the depth dependence of the active intensity vector during each of the three phases. The vertical component of the active intensity is approximated between each pair of neighboring hydrophones by finite difference methods. A numerical approach based on parabolic wave equation simulations [Reinhall and Dahl, J. Acoust. Soc. Am. **130**, 1209 (2011)] is modified here and used to construct theoretical intensity streamlines, motivated by those depicted in Zampolli *et al.* [Acoust. Soc. Am. **133**, 72 (2013)]. These intensity streamlines are validated by the experimental measurements. Energy propagation through the water column and the sediment-water interface is also discussed.

## Session 2aPAb

# Physical Acoustics and Structural Acoustics and Vibration: Phononic Crystals and Metamaterials

Joel Mobley, Cochair
*Phys., Univ. of Mississippi, 1034 NCPA, One Coliseum Dr., University, MS 38677*

Cecille Labuda, Cochair
*National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677*

**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**2aPAb1. Coupled membranes with doubly negative mass density and bulk modulus.** Ping Sheng (Dept. of Phys., Hong Kong Univ. of Sci. & Technol., Clear Water Bay, Kowloon, Hong Kong 000, China, sheng@ust.hk)

We present a structurally and conceptually simple acoustic double negative metamaterial comprising two coupled membranes [Phys. Rev. Lett. **110**, 134301 (2013)]. Owing to its symmetry, the system can generate both monopolar and dipolar resonances that are separately tunable, thereby making broadband double negativity possible. A homogenization scheme is implemented that enables the exact characterization of our metamaterial by the effective mass density and bulk modulus even beyond the usual long-wavelength regime, with the measured displacement fields on the sample's surfaces as inputs. Double negativity is achieved in the frequency range of 520–830 Hz. Transmission and reflection predictions using effective parameters are shown to agree remarkably well with the experiment. Work done in collaboration with M. Yang, G. C. Ma, and Z. Y. Yang.

**8:25**

**2aPAb2. Acoustic cloaking for airborne sound based on inclusions of rigid scatterers.** Victor Manuel Garcia-Chocano, Francisco Cervera, Ana Díaz-Rubio, Alfonso Climente, Daniel Torrent, and José Sánchez-Dehesa (Universitat Politècnica de València, C/Camino de Vera S/N, Departamento de Ingeniería Electrónica, Valencia 46022, Spain, vicgarch@upvnet.upv.es)

Acoustic cloaking is a phenomenon whose physical realization depends on the ability of designing metamaterials with the appropriate parameters. When dealing with airborne sound almost any solid behaves as an acoustically rigid material, so cloaks based on rigid scatterers are here studied. Since these structures are not able to increase the effective sound speed with respect to the background, additional mechanisms should be introduced to allow this effect. Here we will report an acoustic cloak based on temperature gradients. Another possibility of hiding objects from an external sound source consists of using a set of external layers that cancels the scattered field by such objects at selected frequencies. We present the practical realization of this approach for 2D and 3D structures. [Work supported by MINECO from Spain and ONR from United States.]

**8:45**

**2aPAb3. Anomalous ultrasonic transport in phononic crystals with overlapping Bragg and hybridization gaps.** John H. Page, Eric JS Lee, and Charles Croënne (Phys. and Astronomy, Univ. of MB, 301 Allen Bldg., 30A Sifton Rd., Winnipeg, MB R3T 2N2, Canada, john.page@umanitoba.ca)

Many of the interesting properties of phononic crystals are due to the existence of band gaps, which may arise from a number of different mechanisms. These include Bragg scattering as well as hybridization effects, the latter occurring when there are strong scattering resonances that hybridize with a propagating mode of the embedding medium. In this talk, we investigate the interaction between Bragg and hybridization effects on the band gap properties of 2D phononic crystals consisting of nylon rods arranged in a triangular lattice and immersed in water. The lattice constant and rod diameter were chosen to ensure that both mechanisms occur in the same frequency range. The scattering resonances of the nylon rods can be tuned in frequency by varying the temperature, enabling fine control of the overlap between hybridization and Bragg effects. The dispersion relations and transmission coefficient were measured experimentally from the phase and amplitude of transmitted ultrasonic pulses and calculated theoretically by finite element simulations. Strikingly unusual dispersion effects are observed, revealing a novel way of tuning band gap properties. These remarkable characteristics are explained by the competition between two co-existing propagation modes, reflecting strong coupling between scattering resonances of the rods.

9:05

**2aPAb4. An effective Cosserat continuum model for waves in a material with microstructure.** Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu)

The effective acoustic properties of a medium can be tailored by appropriately designing its microstructural components. The effective properties of a material may be generally defined as those that describe the limiting case where the wavelength of propagation is infinite compared to the characteristic scale of the microstructure. Generally, the limit of vanishingly small microstructural scale in an elastic medium results in an effective medium that is again elastic. In this contribution, we give an example for which the above limit results not in an effective elastic medium, but rather, an effective Cosserat medium. We briefly review the properties of Cosserat continua, provide a basic derivation of their fundamental equations of motion, and interpret the continuum properties in terms of the microstructural elements.

9:20

**2aPAb5. Nonlinearity parameters B/A and C/A and wave equation for heterogeneous media containing negative stiffness inclusions.** Stephanie G. Konarski, Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, skonarski@utexas.edu)

This work considers nonlinear propagation in a medium consisting of a low volume fraction of metamaterial inclusions dispersed in a fluid-like material. The metamaterial inclusions of interest are assumed to possess non-monotonic stress-strain constitutive relations, which results in regimes of negative stiffness. For modeling purposes, the constitutive relation for these inclusions is approximated with an expansion to third order in volume strain with coefficients that can be tuned with the geometry of the metamaterial structure and ambient pressure. A far-reaching goal of this research is to model the hysteretic response of the heterogeneous medium resulting from metamaterial inclusion snapping events and the associated effect on acoustic disturbances that cycle through regimes of both positive and negative stiffness. As an initial step, results are presented here for small but finite-amplitude disturbances limited to local regions of the constitutive relation. For this case, the quadratic and cubic nonlinearity parameters B/A and C/A, respectively, as traditionally defined for fluids are obtained. An evolution equation with both quadratic and cubic nonlinearity is also obtained. Numerical solutions of the evolution equation illustrate nonlinear waveform distortion as a function of the volume fraction and constitutive behavior of the inclusions. [Work supported by ARL:UT McKinney Fellowship in Acoustics.]

9:35

**2aPAb6. Two-dimensional broadband acoustic black hole for underwater applications.** Christina J. Naify (National Res. Council PostDC Naval Res. Lab Code 7160, 4555 Overlook Ave SW, Washington, DC 20375, christina.naify.ctr@nrl.navy.mil), Theodore P. Martin (Acoust., Naval Res. Lab Code 7160, Washington, DC), Christopher N. Layman (National Res. Council PostDC Naval Res. Lab Code 7160, Washington, DC), David C. Calvo, and Gregory J. Orris (Acoust., Naval Res. Lab Code 7160, Washington, DC)

Transformation acoustics using sub-wavelength elements to obtain homogenized properties has been utilized increasingly in recent years to manipulate the propagation of acoustic waves. Gradient index (GRIN) lenses are designed by varying local properties of the elements through a given geometry for a variety of devices including focusing lenses and black holes. This study presents a cylindrical, two-dimensional acoustic black hole for underwater applications. The black hole designed focuses acoustic energy to the center of the cylinder using a radially decreasing sound speed profile. An absorbing core then prevents scattering of the acoustic waves into the surrounding fluid. Transformation acoustics was used to design the index gradient through the black hole structure and multiple scattering theory (MST) was used to predict the scattering profile over a broadband range of frequencies. The black hole was constructed of concentric rings of silicone rubber cylinders, which behave as effective fluids in a multiple scattering configuration. Experimental realization of the black hole structure was fabricated and analyzed with measured pressure intensity agreeing with predicted results. [Work sponsored by the Office of Naval Research.]

**9:50–10:15 Break**

10:15

**2aPAb7. A negative stiffness metamaterial inclusion via small-scale curved structural elements.** Timothy Klatt and Michael R. Haberman (Appl. Res. Labs. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

This work presents a means to produce negative stiffness metamaterial units without employing inherent nonlinear material behavior or dynamic effects, but rather through large deformations of tailored microscale geometry. Specifically, the work of Qiu *et al*. [J. Microelectromech. Syst. **13**, 137–146 (2004)] is generalized to create metamaterial units displaying non-monotonic stress-strain relationships with regions of negative stiffness behavior. These cells are designed to be embedded into a continuous matrix material and negative stiffness is elicited through pre-strain of the metamaterial unit cells brought about by changes in ambient pressure. This work presents a nonlinear hierarchical multiscale material model to estimate the macroscopic stiffness and loss of a composite material containing these pre-strained microscale structured inclusions. The multiscale model consists of two scale transition models: (i) an energy-based nonlinear finite element (FE) method to determine the anisotropic tangent moduli of the inclusion, and (ii) an analytical micromechanical model to determine the effective stiffness and loss tensors due to small perturbations about the local strain state of the metamaterial inclusions. Models and results for various volume fractions are presented and discussed.

10:30

**2aPAb8. Negative refraction and focusing of acoustic waves using a foam-like metallic structure.** Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu), Jérôme O. Vasseur, Gérard Haw (Institut d'Electronique, de Microelectronique et de Nanotechnologie, UMR 8520 CNRS, Lille, France), Charles Croënne (Phys. and Mater. Sci., City Univ. of Hong Kong, Hong Kong, Hong Kong), Lionel Haumesser (Groupe de Recherche en Materiaux, Microelectronique, Acoustique et Nanotechnologie, Univ. François Rabelais, Tours, France), and Anne-Christine Hladky-Hennion (Institut d'Electronique, de Microelectronique et de Nanotechnologie, UMR 8520 CNRS, Lille, France)

A phononic crystal (PC) slab made of a single metallic phase is shown, theoretically and experimentally, to display perfect negative index matching and focusing capability in water. The PC slab is a centimeter scale air-filled foam-like metal structure arranged in a regular hexagonal lattice in which the acoustic energy is mediated via the metal lattice. The structure has the density and longitudinal velocity of water at low frequency. The negative index property arises from an isolated higher frequency branch of the dispersion curves corresponding to a mode that couples to incident acoustic waves in water. This band intercepts the sonic line at a frequency in the ultrasonic range, which leads to perfect index matching. The metallic structure is consequently a candidate for the negative refraction of incident longitudinal waves, and the realization of flat superlenses which can focus acoustic waves with a resolution lower than the diffraction limit.

10:45

**2aPAb9. Effective acoustic response and spectral representation.** Raul Esquivel-Sirvent (Instituto de Fisica, UNAM, Apdo Postal 20-364, Mexico DF 01000, Mexico, raul@fisica.unam.mx)

The problem of finding an effective or average acoustic properties of inhomogeneous materials has been treated extensively in connexion with problems of porous media, rock physics and composite materials. Recently, there has been a renewed interest due to the potential applications of acoustic metamaterials. There are different techniques for finding effective acoustic properties. These range from simple averaging techniques, to variational methods, to coherent phase approximations. There is not a unique method for finding effective properties. In the case of dielectric materials, there are at least thirteen procedures reported. In this work, we present the acoustic version of the spectral representation of effective media, first developed to find dielectric effective properties. This method has the advantage that it separates the geometric contribution from the physical property to be calculated, in our case the acoustic impedance. The method is based on a Green's function solution of the acoustic wave equation and finding the effective properties is done by calculating the poles of the so called spectral function. Furthermore, we show that any effective medium model that can be described in the spectral representation satisfies the Kramers-Kronig relations. Numerical examples and comparisons between the spectral representation and other existing procedures will be discussed.

**11:00**

**2aPAb10. Sound attenuation in ducts using locally resonant periodic flush mounted flexible silicon aerogel patches.** Maaz Farooqui, Wael Akl, and Tamer Elnady (Mech. Eng., Ain Shams Univ., ASU GARDS, Abdou basha, Abbasseya, Cairo, Cairo 11517, Egypt, moaz.farooqui@eng.asu. edu.eg)

In recent years, low frequency noise has become an important factor especially in the Aircraft, HVAC, and Automotive industries. In order to reduce this low frequency noise, noise attenuation by the classical Helmholtz resonators has size limitations due to the large wavelengths. Promising noise reductions, with flush mounted Silicon Aerogel patches, can be obtained implementing attenuation due to local resonance and that too without any size constraints. The objective of the current paper is to introduce locally resonant Silicon Aerogel patches flush mounted to an acoustic duct walls aiming at creating frequency stop bands at the low frequency zone (below 500 Hz). Green's Function is used under the framework of interface response theory to predict the degree of attenuation of the local resonant patches. Realistic techniques for expanding the stop bandwidth have been introduced and difference between the Bragg scattering and the locally resonant mechanism was demonstrated using mathematical models. The effect of the arrays of patches on the effective dynamic density and bulk modulus has also been investigated. It is also shown that the numbers and periodicity of these local resonators also plays role in determining the depth and width of the acoustic band gap.

TUESDAY MORNING, 3 DECEMBER 2013 POWELL, 9:00 A.M. TO 11:45 A.M.

## Session 2aSA

## Structural Acoustics and Vibration: Vibration from Aeroacoustic Loads

Micah R. Shepherd, Cochair
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Matthew D. Shaw, Cochair
*Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802*

**Chair's Introduction—9:00**

### *Invited Papers*

**9:05**

**2aSA1. A history of random vibration.** Thomas L. Paez (Thomas Paez Consulting, 185 Valley View Dr., Sedona, AZ 86336, tlpaez4444@gmail.com)

Natural, random, dynamic environments are ubiquitous, and humans have observed them for millennia. Some random dynamic environments are earthquake ground motions, winds, ocean waves, rough roads, and acoustic pressure environments arising from thunder. Before people could easily conceptualize harmonic motions, they observed random, oscillatory environments. Today, random vibration can be mathematically modeled as the motion of a mechanical system excited by a random input. The mathematical theory of random vibration is essential to the realistic modeling of structural dynamic systems. This paper summarizes the work of some key contributors to the theory of random vibration from the time of Rayleigh and Einstein to the present. Among many other things, we describe the works of (1) Rayleigh, who, in the late nineteenth century introduced ideas about the representation of random signals as the superposition of random harmonics; (2) Schuster, who, in the late nineteenth and early twentieth centuries described, without taking its limit, the spectral density; (3) Einstein who, in 1905, wrote the first actual paper on the theory of random vibration (Brownian movement); and (4) Wiener who, in 1930, formally defined the spectral density. Several graphic examples are included.

**9:25**

**2aSA2. Uncertainty bounds estimation when simulating vibration response under aeroacoustic loading.** Paul Bremner (AeroHydroPLUS, 2311 Via Aprilia, Del Mar, CA 92014, pbremner@aerohydroplus.com)

The simulation of linear vibration response under aeroacoustic loading, involves the aeroacoustic loading spectrum and its spatial distribution, and the modal dynamics of the loaded structure. In most practical applications, this problem can only be addressed statistically because the aeroacoustic loading is temporally random and only partially space-correlated, requiring a time-averaged and space-averaged cross-spectrum description of both the loading and the structure vibration response. However, it is equally common that the spectrum and spatial correlation of the loading is uncertain and—particularly at higher frequencies or higher modal densities—even the modal dynamics of the structure are uncertain (e.g., due to uncertainties in the as-built boundary conditions). This paper will explore what is required to predict the statistical variance and the maximum expected vibration response—in addition to the space-time average cross spectrum—given uncertainty in the modeling parameters.

**9:45**

**2aSA3. Measurement of high amplitude relief valve noise during a full scale blowdown.** Neal Evans (Southwest Res. Inst., Div 18, B77, 6220 Culebra Rd., San Antonio, TX 78238, neal.evans@swri.org)

Dynamic pressure fluctuations inside a pipe were measured downstream of a pressure relief valve during a full scale blowdown test. Nitrogen gas flowed with a maximum rate of 33.5 kg/s through a 3 in.x4 in. relief valve generating a peak dynamic pressure level greater than 650 kPa and sustained levels of over 450 kPa (peak). An accurate estimate of valve-generated noise is necessary when predicting radiated noise and acoustic induced vibration, which has been shown to cause fatigue failures at welded discontinuities in piping systems downstream of high pressure drop devices. These failures can be hazardous and costly, particularly when the process involves hydrocarbons such as natural gas. The measured level is compared to existing noise calculation techniques which appear to under-predict the generated noise.



**10:05–10:20 Break**

**10:20**

**2aSA4. Sonic fatigue coherence models.** Robert D. Blevins (United Technologies, 3818 Pringle St., San Diego, CA 92103, rdblevins@AOL.com)

Computational and experimental based models for coherence in acoustic and turbulent pressure fields are developed for application to sonic fatigue of large structures—structures with dimensions greater than a wavelength or of the turbulence length scale. Conservative analytical methods for sonic fatigue of structures exposed to coherent pressure fields are first developed. Then models are introduced to account for finite coherence of turbulent and non-coherent pressure fields that arise in turbojet engines. The computational results are compared with experimental data.

**10:40**

**2aSA5. Reduced order modeling for the skin panels of hypersonic vehicles and nonlinear normal modes.** Matthew S. Allen (Eng. Phys./Eng. Mech., Univ. of Wisconsin-Madison, 535 Eng. Res. Bldg., 1500 Eng. Dr., Madison, WI 53706-1609, msallen@engr.wisc.edu)

The skin panels on concept hypersonic (Mach > 5) vehicles are subjected to intense acoustic and thermal loading. As a result, nonlinear structural dynamic models are needed to predict their response with the required level of accuracy. Vehicles such as this carry a large amount of fuel and so the structure must be very light if the overall vehicle is to meet its performance requirements. This talk will discuss the reduced order modeling frameworks that are being used to create models for these types of structures and discuss how nonlinear normal modes are being used to understand how the structure's response changes with the loading amplitude. Nonlinear modes are also being used to evaluate the reduced order models in order to predict the frequency bandwidth and the range of forcing amplitude over which they will be valid. This approach allows the analyst to develop a considerable level of confidence in the reduced order model without having to compute time responses of the full nonlinear finite element model; time response simulations are far too expensive to be used in practice on the structures of interest.

## *Contributed Papers*

**11:00**

**2aSA6. Low noise blower fan for heating and cooling applications.** Yi Liu (Ingersoll Rand, 618 Davis CT, Indianapolis, IN 46234, yiliu1975@gmail.com) and Percy Wang (Ingersoll Rand, Tyler, Texas)

A new low noise fan called foam blower fan is presented here for low noise heating, ventilation, and air conditioning (HVAC) applications. Traditional HVAC blower fans are made of discrete metal or plastic fan blades, which generate blade tonal noise when each blade interacts with blower cut-off or other housing structures. It is proposed here that foam wedge (porous media) is inserted between blades at strategic locations, to reduce the tonal noise generation. It is analytically and experimentally demonstrated in this paper that the proposed new foam fan can reduce discrete tonal noise at various operating speeds.

**11:15**

**2aSA7. Industry tubing characterization method for fixed speed applications.** Yi Liu (Ingersoll Rand, Indianapolis, IN) and Yufeng Shen (Ingersoll Rand, 9/11F Tower B City Ctr. of Shanghai, No. 100 Zun Yi Rd., Shanghai, China, MichaelYuFeng.Shen@trane.com)

Efforts for structural tubing characterization are presented here for HVAC industry fixed speed applications. Structural tubing is a type of metal connector typically with hollow circular or rectangular cross-sections. Copper tubing is widely used in HVAC industry to transfer refrigerant between compressors and condensers. However, when the natural frequencies of tubing system are at the running speeds of the compressors; design life of the product is reduced. Therefore, tubing configuration structural change is needed to shift these problem frequencies. Traditionally, industry has been using finite element method and vibration tests to identify the critical geometric parameter for frequency shift. New efforts are been proposed here to present designer a sensitivity study tool to identify these critical parameters quickly, therefore, shorten the design cycle time.

11:30

**2aSA8. Modification of the spectral response of a pipe resonator using a subordinate array of coupled Helmholtz resonators.** Aldo A. Glean, Joseph F. Vignola, John A. Judge, and Teresa J. Ryan (Mech. Eng., Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, 10glean@cardinalmail.cua.edu)

The dynamic response of a resonant system can be manipulated by attaching a set of substantially smaller resonators with a prescribed distribution of properties. These smaller resonators are collectively referred to as a subordinate array. This work describes an experimental demonstration of such a manipulation, in which we alter one resonant peak of the primary system to create a flat bandpass response, while leaving the other resonances unaffected. In this instance, the primary resonant system is a pipe with one end closed and the other open, and the subordinate array consists of a set of small Helmholtz resonators. We demonstrate the desired modification of the system response at the third resonant frequency without significant change in adjacent resonances. Specifically, a single system resonance of the primary resonator shows a peak response when no subordinates are attached and exhibits a band pass response when the subordinates are attached. Sensitivity of system response to errors in design parameters is discussed and experimental results are compared to theoretical predictions.

TUESDAY MORNING, 3 DECEMBER 2013 — PLAZA B, 8:30 A.M. TO 12:00 NOON

# Session 2aSC

## Speech Communication: Speech Perception I

Grant L. McGuire, Chair
*Linguistics, UC Santa Cruz, 1156 High St., Stevenson Academic Services, Santa Cruz, CA 95064*

### *Contributed Papers*

8:30

**2aSC1. Perception of acoustically similar vowels from English and Hebrew.** Richard J. Morris (Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham Rd., 612 Warren Bldg., Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu) and Yonit A. Shames (Audiol., Helix Hearing Care, Orlando, FL)

The formant theory suggests that vowels are differentiated perceptually based on the presence of concentrated bands of harmonic energy in the vowel's acoustical signal. Vowels from different languages often contain formants with similar configurations, but it is unclear what effect cross-language variations have on the perception of the vowels. This study was completed to find when English-speaking listeners are able to differentiate between vowels spoken in two languages, General American English and Modern Hebrew. Natural vowels in kVp syllables were recorded and acoustically normalized using the Bark scale. The English and Hebrew vowels were paired in test groups based on the normalized formant values and the fundamental frequencies of the speakers. Listeners designated each pair as "same" or "different." It was hypothesized that listeners were more likely to differentiate between the vowels in English-Hebrew pairs than in same-language pairs, and that they were more likely to differentiate between vowels with higher Bark scale differences. However, Bark scale differences did not always match perceived differences. Listeners demonstrated the least difficulty discriminating between different-language pairs containing /e/ and /o/ (60–90% accuracy), more difficulty for English /ɑ̃/ and Hebrew /a/ (40–80% accuracy), and the most difficulty for /i/ and /u/ (10–60% accuracy).

8:45

**2aSC2. Acoustic cue weighting across modalities in a non-native sound contrast.** Jessamyn L. Schertz (Dept. of Linguist, Univ. of Arizona, Douglass 200, Tucson, AZ 85721, jschertz@email.arizona.edu), Andrew J. Lotto (Dept. of Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ), Natasha Warner (Dept. of Linguist, Univ. of Arizona, Tucson, AZ), and Taehong Cho (Hanyang Phonet. and PsychoLinguist Lab, Dept. of English Lang. and Lit., Hanyang Univ., Seoul, South Korea)

This work investigates the production and perception of the English stop voicing contrast on multiple acoustic dimensions by native speakers of Seoul Korean. Subjects completed a production task as well as a forced-choice identification task on stimuli varying on three acoustic dimensions (aspiration duration, pitch, and closure duration) in both English and Korean. On average, native Korean listeners relied more on pitch than on aspiration duration to categorize the English stop voicing contrast (which native English listeners distinguish primarily by aspiration). However, individual categorization patterns differed considerably, with some listeners using only pitch, some using only aspiration, and most using both. In contrast, in production, aspiration duration was a better predictor of voicing category than pitch, although pitch was still a stronger predictor than previously found in native English productions. The heavier reliance on pitch by Korean listeners may be attributable to the greater importance of pitch in their native stop contrast; however, there does not appear to be a consistent, straightforward mapping of English sounds onto the Korean categories. Results will be discussed in the terms of the perception-production interface on the level of individual acoustic cues, as well as the influence of native language cue weights on non-native sound contrasts.

9:00

**2aSC3. Perception of stressed vs unstressed vowels: Language-specific and general patterns.** Priscilla Shin, Natasha L. Warner, Maureen Hoffmann (Linguist, Univ. of Arizona, Box 210028, Dept. Ling, Univ. AZ, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), James McQueen (Donders Inst., Radboud Univ. of Nijmegen, Nijmegen, Netherlands), and Anne Cutler (MARCS Inst., Univ. of Western Sydney, Penrith, NSW, Australia)

Unstressed vowels are somewhat centralized (even full vowels such as the second in "city, taco"), reducing their acoustic distinctiveness. The current work compares listeners' perception of stressed and unstressed vowels in English and Dutch. The data come from two large projects on native listeners' perception of all possible diphones (CV, VC, CC, and VV sequences, all vowels stressed and unstressed) in English and Dutch. These datasets provide information about listeners' uptake of perceptual cues over time that is comparable across the two languages. Both groups perceived unstressed vowels less accurately than stressed, but this effect was far larger for English. English listeners showed a very large stress effect for lax vowels and a moderate effect for other vowels, while the Dutch listeners showed effects that were small and largely restricted to /a/. Dutch listeners may be able to identify unstressed vowels better than English listeners because the stressed-unstressed distinction has more informational value in Dutch than in English. However, both languages showed a larger stress effect just before a following consonant. This suggests that consonantal coarticulation obscures the quality of unstressed vowels in both languages. Thus, perception of stressed vs unstressed vowels demonstrates both language specificity and cross-language commonality.

**9:15**

**2aSC4. Phonetic and orthographic cues are weighted in speech sound perception by second language speakers: Evidence from Greek speakers of English.** Anastasia Giannakopoulou (Psych., Univ. of Bedfordshire, University Square, Luton LU1 3JU, United Kingdom, anastasia.giannakopoulou@beds.ac.uk), Maria Uther (Psych., Winchester Univ., Winchester, United Kingdom), and Sari Ylinen (Psych., Univ. of Helsinki, Helsinki, Finland)

Speech-sound contrasts that have multiple phonetic cues can be particularly difficult for foreign-language learners especially if the cues are weighted differently in the foreign and native languages (e.g., Giannakopoulou *et al.*, 2013). The orthographic representation of words is suggested to also interfere with speech sound perception in way of presenting additional cues for the second language learner. In order to examine the possibility that orthographic representation of the word stimuli provides additional cues, this study explores perceptual identification with the use of pictures as visual stimuli. Greek child and adult speakers of English were studied to determine on what basis they are making perceptual identification between English vowels. One task involved the use of minimal pairs in their orthographic form (word stimuli), another task used relevant pictures that resembled the meaning of the respective words. The auditory stimuli used in both task types were identical. Performance was impaired for Greek speakers across all tasks but worst for Greek speakers for the picture stimuli task. Interestingly, child Greek speakers performed significantly worse in the picture stimuli task, even though a picture translation control task revealed high performance. These results are discussed in terms of the strategies used to acquire new languages.

**9:30**

**2aSC5. Evidence for cognitive restoration of time-reversed speech by a language-trained chimpanzee (*Pan troglodytes*).** Lisa A. Heimbauer (Psych., Penn State Univ., 442 Moore Bldg., University Park, PA 16802, lisa.heimbauer@gmail.com), Michael J. Beran (Lang. Res. Ctr., Georgia State Univ., Atlanta, GA), and Michael J. Owren (Psych., Emory Univ., Atlanta, GA)

Previously, we reported on the ability of Panzee, a language-trained chimpanzee, to identify sine-wave and noise-vocoded speech by attending to the amplitude and frequency modulations in the altered signals. Here, we report on her ability to perceive phoneme-length information in words reproduced in time-reversed form. While this manipulation preserves the amplitude of frequency components, it reverses the pattern of energy changes within each reversal window. Listeners easily recognize speech at reversal windows up to 100-ms length, but at longer reversal lengths unintelligibility begins to occur (Saberi and Perrott, 1999). The theoretical interpretation is that individual phonetic segments range from 50 to 100 ms (Crystal and House, 1988), and reversal-windows less than 100 ms provide for restoration of phoneme perception. Hypothesizing that Panzee also perceives speech based on phonemic segments, we tested her and humans with words in eight reversal forms ranging from 25 to 200 ms. Results revealed time-reversal window length significantly predicted percentage-correct word identification for Panzee and the humans. Additionally, window lengths exceeding 100 ms produced partial word intelligibility, with 50% intelligibility occurring at approximately 130 ms for both species. We therefore conclude that Panzee attends to phoneme-related cues in time-reversed speech, and hence in natural speech.

**09:45–10:00 General Discussion**

**10:00–10:30 Break**

**10:30**

**2aSC6. Simple auditory elements induce perception of a phonetic feature.** Gregory Finley (Linguist, Univ. of California, 5820 Occidental St., Oakland, CA 94608, finley@berkeley.edu)

In this presentation, I demonstrate that certain nonspeech sounds can have perceptual phonetic value. I focus on a single phonetic/articulatory feature, lip rounding, and its detection in simple auditory stimuli. Behavioral experiments show that a rounding percept is possible for two types of nonspeech. One stimulus type, which yields the more robust response, is a complex periodic source filtered by a single narrow band reminiscent of a speech formant. The resulting nonspeech varies in perceived roundedness depending on the filter's frequency, corresponding roughly with F2. The other stimulus type is a pure tone modulated upward in frequency. Preliminary results suggest that rounding can indeed be perceived on these sounds, but only with specific modulation rates within a certain frequency range. These findings indicate that minimally simple auditory objects, including pure tones and filtered bands, can be sufficient to encode phonetic information. Additionally, these two types of cues diverge in their ability to trigger this percept: a filtered band works as a static spectral cue, whereas a pure tone requires spectrotemporal modulation. This observation is consistent with findings that there are auditory STRFs specifically sensitive to modulation and the theoretical perspective that auditory organization directly predicts the processing of speech.



**10:45**

**2aSC7. The effect of speaking rate, vowel context, and speaker intelligibility on the perception of consonant vowel consonants in noise.** Anirudh Raju and Abeer Alwan (Elec. Eng., Univ. of California, Los Angeles, 550 Veteran Ave., Apt. #102, Los Angeles, CA 90024, anirudh90@ucla.edu)

In this paper, we perform pilot experiments to evaluate the feasibility of a model to predict human recognition of speech sounds in the presence of noise at different speaking rates. CVC stimuli comprising a phonetically balanced set of 13 consonants and 3 vowels (/i/, /a/, /u/) were recorded in a sound proof booth by two talkers at two different speaking rates (fast and slow). Noisy stimuli were generated by adding babble noise at different levels to the quiet recordings. These stimuli were used to conduct perceptual experiments in which listeners were asked to listen and repeat back the CVC phrases presented in babble noise under 3 SNR conditions and both speaking rates. The data were transcribed by two trained linguists. Consonant confusion matrices were generated from these data and were analyzed by noise level, speaker, center vowel, and speaking rate. With the exception of /CuC/ stimuli, speaking rate had the most pronounced effect on perception with slow speech being more intelligible than fast speech in noise. /CaC/ stimuli were, on average, more robust than other stimuli in all conditions and one talker was significantly more intelligible than the other. A detailed analysis of the results will be presented. [Work supported in part by the NSF.]

**11:00**

**2aSC8. Perceptual compensation with familiar and unfamiliar rounding coarticulation.** Keith Johnson, Shinae Kang, and Emily Cibelli (Dept. of Linguist, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720-2650, keithjohnson@berkeley.edu)

We compared the integration of three kinds of contextual information in the perception of the fricatives [s] and [ʃ]. We asked American English listeners to identify sounds on an [s] to [ʃ] continuum and manipulated (1) the vowel context of the fricative ([Ce], [Co], [Cœ]), (2) the original fricative of the CV ([s] vs [ʃ]), and (3) the modality of the stimulus (audio-only, AV). There was a large compensation for coarticulation effect on perception—subjects responded with "s" more often when the following vowel was round. Interestingly, and perhaps significantly, perceptual compensation was not as great with the less familiar vowel [œ] even when listeners saw the face. Measurements of lip rounding in these stimuli show that [o] and [œ] have about the same degree and type of rounding over the CV. In a second experiment, we measured reaction time to audio-visual mismatches in these stimuli (again in a fricative identification task). We found that mismatches of audio and video consonant information slowed reaction time, and that vowel mismatches did as well. However, mismatch between [o] and [œ] did not slow reaction time. These data suggest that linguistic experience and stimulus properties affect perception.

11:15

**2aSC9. Visual cue salience depends on place of articulation and syllabic context.** Shinae Kang and Keith Johnson (Linguist, UC Berkeley, 1203 Dwinelle Hall, UC Berkeley, Berkeley, CA 94720-2650, sakang2@berkeley.edu)

This study is on audio-visual perceptual intelligibility of consonants in intervocalic clusters (VC1C2V). Previous studies have yielded inconsistent findings on perceptual salience of different stop consonants and very few have tested salience in clusters. Consequently, it has been unclear as to whether greater or less perceptual salience leads to greater degree of place assimilation. In Korean, labials are often produced with more gestural overlap than velars in C1. I tested whether labials are perceptually more or less salient in both audio and audio-visual conditions. VC and CV syllables spoken by both English and Korean speakers were first embedded in noise and spliced together for non-overlapping VCCV sequences. Korean listeners identified the two consonants in either audio or AV presentations. A confusion matrix analysis for each stop consonant shows that in C1 there is asymmetric improvement with the addition of videos for labial consonants only, while in C2 this asymmetry was not found. The result suggests that listeners make differential use of visual cues depending on place of articulation and syllabic context. Also, the result supports the talker enhancement view of sound change, which assumes that talkers are aware of perceptual salience and enhance (with less gestural overlap) the weak contrast.

11:30

**2aSC10. Psychophysiological indices of effortful listening in younger and older adults.** Alexander L. Francis (Speech, Lang. and Hearing Sci., Purdue Univ., SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu), Megan K. MacPherson (School of Commun. Sci. and Disord., Florida State Univ., Talahassee, FL), Bharath Chandrasekaran (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX), Ann M. Alvar (Speech, Lang. and Hearing Sci., Purdue Univ., West Lafayette, IN), and Fernando Llanos (School of Lang. and Cultures, Purdue Univ., West Lafayette, IN)

Older adults often have difficulty understanding speech in background noise, and this difficulty may be associated with cognitive processing demand. According to the effortfulness hypothesis, even sub-clinical age-related changes in hearing may increase cognitive demand for speech understanding, making listening in noise more effortful for older adults even when recognition performance is comparable to that of younger listeners. Separating speech from background noise requires both segregating target from masking signals and selectively attending to the target while ignoring maskers. While both segregation and selection may demand cognitive resources, it is not known whether both mechanisms interact with age to the same degree. To address this question, younger and older adults listened to and repeated sentences presented in quiet and under conditions that put relatively more emphasis on segregation (energetic masking using speech-shaped broad-band noise) or selection (informational masking using two-talker babble) or are cognitively demanding without masking (synthetic speech). Masked stimuli were equally intelligible based on prior research, so differences in listening effort may be attributed to age and/or masker type. Listening effort was measured behaviorally via traditional rating scales (NASA TLX), and psychophysiologically in terms of autonomic nervous system responses (heart rate, pulse period, and amplitude, and skin conductance).

**11:45–12:00 General Discussion**

TUESDAY MORNING, 3 DECEMBER 2013 — CONTINENTAL 4, 8:30 A.M. TO 11:30 A.M.

## Session 2aSP

## Signal Processing in Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Annual Buacoustics: Time Reversal for Localization and Focusing of Sound

Brian E. Anderson, Chair
*Geophys. Group, Los Alamos National Lab., M.S. D443, Los Alamos, NM 87545*

### *Invited Papers*

8:30

**2aSP1. Time-reversal-based underwater acoustic communication.** Hee-Chun Song (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, hcsong@mpl.ucsd.edu)

Time reversal (TR) exploits spatial diversity to achieve spatial and temporal focusing in complex environments. Over the last decade its concept has been applied successfully to underwater acoustic communications in time-varying multipath environments with robustness and computational simplicity, as an alternative to conventional adaptive multichannel equalization. Temporal focusing (pulse compression) mitigates the intersymbol interference (ISI) and subsequent channel equalization removes the residual ISI, thus providing nearly optimal performance in theory. The spatial focusing capability facilitates multi-user communications without an explicit use of time, frequency, or code division, while an adaptive time reversal approach can further reduce the crosstalk among users. TR communications can be easily extended to time-varying channels using a block-based approach with periodic channel updates. This talk will present an overview of TR communications and recent advances including bidirectional equalization, single- versus multi-carrier approach, and communications with autonomous vehicles such as an AUV or Glider.

**8:50**

**2aSP2. Time reversal methods for high frequency multiple-input/multiple-output underwater acoustic communication.** Aijun Song and Mohsen Badiey (School of Marine Sci. and Policy, Univ. of Delaware, 114C Robinson Hall,Newark, DE 19716, ajsong@udel.edu)

As a sound focusing technique, time reversal has been utilized in underwater acoustic communication since the 1990s. Here, we particularly study the usage of time reversal methods to separate sounds from different sound sources for high frequency transmissions (greater than 10 kHz), in addition to sound focusing. The separation of sounds creates multiple parallel acoustic links between multiple transducers and a receiver array, thus, providing increased data rates between the source and receiver for communication purposes. It is referred to as multiple-input/multiple-output (MIMO) acoustic communication. Multiple at-sea experiments have been conducted to demonstrate high frequency MIMO signaling and their communication performance. A number of signal processing techniques have been developed in the time reversal framework to enhance the sound focusing and separation that are needed in high frequency acoustic MIMO communication. For example, multi-stage interference cancellation methods have been devised to suppress the co-channel interference and to ensure the sound separation. In this talk, we will show their effectiveness in the presence of water column variability based on our experimental data. An acoustic communication channel model will also be used to explain the impact of environmental variability.



**9:10**

**2aSP3. Time reversal communication over doubly spread channels.** Wen-Jun Zeng (Dept. of Electron. Eng., City Univ. of Hong Kong, Kowloon, Hong Kong 100084, Hong Kong, cengwj06@mails.tsinghua.edu.cn) and Xue Jiang (Dept. of Elec. and Comput. Eng., McMaster Univ., Hamilton, ON, Canada)

Conventional time reversal can mitigate multipath delay dispersion by temporal focusing. But it is not applicable to time-varying channels with a Doppler spread. Although recently time reversal communication has been adapted to time-variant channels, the modified technique requires frequent channel updates to track channel variations and cannot handle large Doppler spread, which means that it cannot achieve frequency focusing. In this paper, two time reversal receivers for underwater acoustic communications over doubly spread channels are proposed. The proposed approach, which can be interpreted as time-frequency channel matching, is based on the channel spreading function rather than impulse response adopted by the existing techniques; this leads to much less frequent channel updates. Unlike existing methods that only correct a single Doppler shift, the proposed approach uses a rake-like structure to compensate for multiple Doppler shifts and hence can eliminate severe Doppler spread induced by temporal channel variations. Simulation results verify the effectiveness of the proposed approach, indicating that it can simultaneously counteract delay and Doppler spreads, achieving both temporal and frequency focusing.

**9:30**

**2aSP4. Advances in biomedical applications of time reversal acoustic focusing of ultrasound.** Alexander Sutin (, Artann Labs., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu), Yegor Sinelnikov, and Armen Sarvazyan (, Artann Labs., West Trenton, NJ)

Time reversal acoustics (TRA) is one of the most efficient methods of ultrasound focusing in heterogeneous composite biological media, especially inside reverberating cavities, such as the skull. In this talk, we will overview several recently developed therapeutic applications of TRA focusing of ultrasound including enhanced drug delivery to brain tumors, generation of focal regions of complex shape tailored to the geometry of the target lesion and TRA dynamic focusing, that allows to maintain the constant acoustic intensity in the focus regardless the variation of acoustical parameters of the media. We are currently developing several new medical applications of TRA, such as remote charging of batteries in internal implants and leadless energizing deep brain stimulators, which are based on the possibility to remotely generate an electrical signal in tissue using TRA principles. These applications employ a TRA focusing system with wireless electromagnetic feedback from the tiny implanted piezotransducer acting as a beacon. The acoustic energy is accurately focused at the piezotransducer generating required electrical signal while providing minimum exposure of surrounding tissues to ultrasound energy. Possibility of remote generation of electrical signals in tissue with amplitudes reaching tens of volts was demonstrated. [Work supported by NIH R21 CA164935-01.]

**9:50**

**2aSP5. Invariants of the time reversal operator and characterization of solid media: An overview.** Claire Prada (Institut Langevin, CNRS ESPCI, 1 rue Jussieu, Paris 75005, France, claire.prada-julia@espci.fr)

The invariants of the time reversal operators for solid media have been the object of several studies in the past 15 years in the context of multi-element array imaging. These invariants are obtained from the decomposition of the array response matrix. Their analysis was applied to the detection of flaws using bulk waves, or Rayleigh and Lamb guided more. It was also applied to the characterization of shells using the radiation of circumferential guided modes. A review of the different solutions that have been proposed to improve this analysis will be given. Then, selected experimental examples of time reversal invariants will be discussed.

**10:10–10:25 Break**

**10:25**

**2aSP6. Evaluation of concrete carbonation using time reversal and nonlinear acoustics.** Pierre-yves Le Bas (Geophys. group, EES-17, Los Alamos National Lab., MS D446, Los Alamos, NM 87545, pylb@lanl.gov), Cedric Payan (Aix Marseille Univ., LMA CNRS UPR 7051, Marseille, France), Timothy J. Ulrich (Geophys. group, EES-17, Los Alamos National Lab., Los Alamos, NM), and Vincent Garnier (Aix Marseille Univ., LMA CNRS UPR 7051, Aix, France)

Carbonation of concrete can lead to corrosion of rebar and degradation of structures, including nuclear plant walls and reactors and casks for storage of nuclear waste. Carbonation has been shown to decrease the natural nonlinearity of concrete. Using the time reversal nonlinear elastic diagnostic (TREND) at different frequencies allows for probing at different depths (half a wavelength). By looking at

the evolution of the nonlinear response with frequency we can estimate the depth of carbonation. This study will present experimental results aimed at determining the depth of carbonation on medium scale samples ($25\times50\times10$ cm and $75\times100\times20$ cm). The samples have been prepared using a protocol known to induce carbonation down to a controlled depth. Several samples are available with carbonation at 0, 1, 2, and 4 cm. We will present the results of the experiments analyzed using several techniques to quantify nonlinearity, namely, pulse inversion and Scaled Subtraction Method (SSM).

## *Contributed Papers*

**10:45**

**2aSP7. Applying an old appealing idea to modern seismology: Time reversal to characterize earthquakes.** Carene Larmat, Robert A. Guyer, Paul A. Johnson (EES-17, Los Alamos National Lab., P.O. Box 1663, M.S. D452, Los Alamos, NM 87545, carene@lanl.gov), and Jean-Paul Montagner (Seismology Lab., Institut de Physique du Globe de Paris, Paris, Paris Cedex 05, France)

Wave physics is one domain where reversing time is possible and has led to interesting applications. In acoustics, Parvulescu and Clay (1965) used what they termed a "matched signal technique" to beat multi-reverberation in the shallow sea. In seismology, McMechan (1982) demonstrated the feasibility of what he termed "wavefield extrapolation" to locate seismic sources. Since then, other concepts and applications, all related to time-reversal, have often been proved to be successful where other techniques have failed. This success is due to the inherent ability of time-reversal to function well in complex propagation media as well as the remarkable robustness of the method with sparse receiver coverage. The key aspect of time-reversal for future applications in seismology is that it relies on no a priori assumption about the source. This allows automatic location of earthquakes and the study of seismic events for which the assumption of point source breaks down. This is the case of big earthquakes (Mw >8) for which the rupture length and source duration extend to hundreds of kilometers and several tens of seconds. We will show an application to the 2011 Japan earthquake, to icequakes related to glaciers motions in Greenland and to seismic tremor with no clear onset.

**11:00**

**2aSP8. A high amplitude non-contact acoustic source: from a proof of concept to the understanding of the mechanisms involved.** Marcel C. Remillieux, Brian E. Anderson, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., Los Alamos, NM 87545, mcr1@lanl.gov)

Recently, the preliminary design of a high-amplitude, non-contact acoustic source was proposed for nondestructive testing applications [Le Bas *et al*., J. Acoust. Soc. Am. **134**, EL52 (2013)]. The design is based on the principle of time reversal, a process to focus energy at a point in space. In the present work, the main physical mechanisms involved in the operation of this device are examined numerically using the finite-element (FE) method. First, a three-dimensional FE model of the device is validated in the frequency domain against experimental data. Subsequently, two-dimensional transient simulations are used to conduct a parametric study on the effect of wall thickness and transducer density on the efficiency of the device. Last, a time reversal numerical experiment is presented. Results from this study can be used to design a more efficient non-contact source. [Work supported by Institutional Support (LDRD) at the Los Alamos National Laboratory.]

**11:15**

**2aSP9. Imaging the orientation of stress corrosion cracking using the three component time reversed elastic nonlinearity diagnostic.** Brent O. Reichman, Brian E. Anderson, Timothy J. Ulrich, and Pierre-Yves Le Bas (Geophys. Group (EES-17), Los Alamos National Lab., M.S. D446, Los Alamos, NM 87545, bea@lanl.gov)

Linear acoustic techniques often are not able to locate closed cracks as the acoustic waves pass right through them. Fortunately, nonlinear acoustic techniques may be used in conjunction with time reversal techniques to not only locate a closed crack but also to image it. This presentation will discuss how cracks may be imaged using the three component time reversed nonlinearity diagnostic (3D-TREND). Specifically, the orientation of individual cracks in a 304L stainless steel plate, which resulted from controlled stress corrosion cracking (SCC) experiments, will be presented along with other imaging information about the crack. 3D-TREND is used to create an individual time reversal focus of energy at each inspection point of interest. The use of different frequencies and different excitation signals has been used in an attempt to extract more information about the crack. [This work was supported by the U.S. Dept. of Energy, Fuel Cycle R&D, Used Fuel Disposition (Storage) Campaign].

## Session 2aUW

# Underwater Acoustics: Sound Propagation Through and Scattering by Internal Waves, Spice, and Finestructure in Shallow Water I: Past, Present, and Future

Steven I. Finette, Cochair
*Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320*

James Lynch, Cochair
*Woods Hole Oceanogr., M.S. # 11, Bigelow 203, Woods Hole, MA 02543*



**Chair's Introduction—8:55**

## *Invited Papers*

**9:00**

**2aUW1. Future shallow water low frequency (100–1000 Hz) acoustic signal propagation physics studies.** Marshall H. Orr (The Acoust. Div., The Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, rubyspiral@gmail.com)

The measurement and prediction of the phase and amplitude properties of acoustic signals (1 to 1000 Hz) propagating in the dynamic shallow water (30 to 500 m water depths) environment has been a focus of the underwater acoustic research community for nearly 20 years. The majority of the ocean studies have occurred during late spring to summertime oceanic conditions when the sound speed variability was influenced by nonlinear mode 1 internal wavefields. Sound speed variability during the summer-to-winter and winter-to-summer water column transition periods will cause variability in the phase coherent properties of acoustic signals. Few experimental studies explicitly focused on quantifying the variability of phase coherent properties of acoustic signals during these periods have been performed. Illustration of the types of fluid processes that will perturb the sound speed structure during the fall-to-winter transition periods will be presented. Included will be mode 1 and mode 2 internal wave perturbation of the sound speed field as well as the locations and spatial scales of sound speed variability caused by interleaving water masses of varying temperature and salinity (Spice). Fall-to-winter transition sound speed fields will be contrasted with summer time sound speed conditions. [Work supported by the Office of Naval Research.]

**9:20**

**2aUW2. Statistics of internal waves measured during the Shallow Water 2006 experiment.** Mohsen Badiey, Lin Wan (College of Earth, Ocean, and Environ., Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu), and James F. Lynch (Woods Hole Oceanographic Inst., Woods Hole, MA)

During the Shallow Water Acoustic Experiment 2006 (SW06), detailed measurements of the time-varying ocean environment were made while simultaneously acoustic signals were transmitted between various source and receiver pairs. The time-varying environment induced by internal waves (IW) was recorded by an array of moored thermistor chains, as well as by the attending research vessels. Using a mapping technique described by Badiey *et al*. [J. Acoust. Soc. Am. EL. **134** (2013)], the three-dimensional (3D) temperature field for over a month of IW events was reconstructed. The results of this mapping are used for the statistical analysis of the IW parameters, such as the IW propagation speed, direction, amplitude, coherence length, etc. This paper provides a summary of these results and also examines the implications of the detailed statistics as regards to the acoustic field. The results in this paper could be used as a database for studying the IW generation, propagation, and its impact on the 3D acoustic propagation in waveguides. [Work supported by ONR322OA.]

## *Contributed Papers*

**9:40**

**2aUW3. A perspective of modeling internal wave/acoustic wave interactions at the Naval Research Laboratory 1992-2012.** Steven I. Finette (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

Interest in modeling internal gravity waves in littoral regions was stimulated by the discovery of anomalous transmission loss in the Yellow Sea, predicted by Zhou and Rogers in 1991 to be caused by solitary wave propagation along the acoustic transmission path. Over the past 20 years, we have been modeling acoustic wave/solitary internal wave interactions and an overview and perspective of some of this work, illustrated by examples, will be presented. The relationship between internal waves and the scintillation index, horizontal refraction of acoustic energy, oceanographic waveguide focusing, mode coupling between acoustic and internal wave modes, and the effect of internal waves on acoustic field uncertainty are discussed. Some thoughts on the future of 4-D simulation of acoustic/internal wave interactions and modeling in the context of incomplete environmental knowledge are considered. [Work supported by the Office of Naval Research.]

**9:55**

**2aUW4. Acoustic normal mode fluctuations due to internal waves in the Shallow Water 2006 experiment.** Lin Wan, Mohsen Badiey (College of Earth, Ocean, and Environ., Univ. of Delaware, 003 Robinson Hall, Newark, DE 19716, wan@udel.edu), and James F. Lynch (Woods Hole Oceanographic Inst., Woods Hole, MA)

The Shallow Water 2006 (SW06) experiment was a large-scale acoustic experiment conducted on the New Jersey continental shelf in the summer of 2006 with substantial investment from the Office of Naval Research. The main goal of this experiment was to find a detailed understanding of the waveguide during the propagation of broadband acoustic signals in the presence of internal waves (IW). One month of IW events recorded during this experiment has been reconstructed with the aids of densely deployed thermistor strings and ship-borne X-Band radars. The comprehensive IW measurements enabled the study of the correlation between the IW-induced time-varying environment and the acoustic normal mode fluctuations. During this experiment, an L-shaped hydrophone array was moored inside the area with IW measurements. Acoustic sources transmitting for 7.5 min every 30 min starting on the hour were deployed at the shallow end of the across-shelf path and the outer end of the along-shelf path respectively. The acoustic modal fluctuations in modal arrival time, intensity, temporal coherence, and spatial coherence during the aforementioned IW events are analyzed. The relationship between the modal behavior and IW parameters is investigated and possible IW inversion schemes using acoustic measurements are discussed. [Work supported by ONR322OA.]

**10:10–10:25 Break**

**10:25**

**2aUW5. Simple expressions for the horizontal array coherence length in shallow water acoustics.** James Lynch, Arthur Newhall, Timothy Duda (Dept. Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., M.S. # 11, Bigelow 203, Woods Hole, MA 02543, jlynch@whoi.edu), William Siegmann (Dept. Appl. Mathematics, Rensselaer Polytechnic Inst., Troy, NY), and John Colosi (Dept. Oceanogr., Naval Postgrad. School, Monterrey, CA)

The use of simplified “feature models” (geometric idealizations of specific, isolated ocean features) for coastal oceanographic features can allow one to calculate acoustically useful quantities approximately and even generate analytic forms for them. Feature models for coastal fronts, eddies, internal tides, linear and nonlinear internal waves, and spice are presented and the scattering of sound from these objects is calculated. This allows one to estimate the useful quantity $L_{coh}$, the horizontal coherence length that represents a physical limit for array signal processing. Calculations of $L_{coh}$ and their comparisons with data will be presented. The effects of acoustic multipath propagation, which can compete with the medium scattering in the estimates of $L_{coh}$, are also estimated using basic models. [Work sponsored by the Office of Naval Research.]

**10:40**

**2aUW6. Impact of fine-scale sound-speed fluctuations on acoustic autocorrelation times in the East China Sea.** Peter C. Mignerey and Altan Turgut (Acoust. Div., Naval Res. Lab., Peter Mignerey Code 7160, Washington, DC 20375-5350, peter.mignerey@nrl.navy.mil)

Autocorrelation times of acoustic signals propagating through shallow oceans are largely driven by sound-speed fluctuations. In August 2008, the Transverse Acoustic Variability Experiment obtained measurements in the East China Sea (65–80 m water depth) of fluctuating signals propagating 33 km from a moored source to a bottomed line array. Supporting environmental measurements were obtained by a towed conductivity-temperature-depth chain. For time periods without large nonlinear internal waves, the measured internal-wave power spectrum shows excess energy at high wavenumbers in comparison with the shallow-water internal-wave model of Levine. Likewise the associated sound-speed fluctuation spectrum exhibits high-wavenumber components in excess of a simple power law. Autocorrelation times of measured 300 Hz acoustic signals were compared with simulated times obtained using a parabolic-equation model to propagate acoustic fields through sound-speed fluctuations driven by linear internal-wave displacements. Results of the comparison are that median measured autocorrelation times (115 s) are shorter than simulations (300 s) driven by spectra with depleted high-wavenumber components. Simple frozen ocean simulations that translate the environment at 0.6 m/s produce autocorrelation times close to the data. [Work supported by the Office of Naval Research.]

**10:55**

**2aUW7. Time reversal of modal arrivals for broadband signals in horizontally stratified shallow water due to internal waves.** Mohsen Badiey and Jing Luo (College of Earth, Ocean, and Environ., Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Shallow water waveguides in the presence of internal waves can make a time-varying, horizontally stratified medium that can significantly affect broadband pulse propagation. In this paper, we analyze a unique data set obtained during the passage of an internal wave event in a region where a broadband sound source (270 to 330 Hz) was received by an L-shaped hydrophone array about 20 km away in 80 meters on the New Jersey continental shelf. During the time 20:30 and 22:07 GMT, on 17 August 2006, an approaching IW affected the dispersion characteristic of the broadband LFM chirp signal while passing an acoustic track in shallow water waveguide. Modal behavior is examined before and after the internal wave front crossed the source-receiver track. While dispersion characteristics of the signals changed, modal arrival time reversal occurred. The corresponding group and phase velocities that signals experienced during this phenomenon are analyzed using the theory of horizontal rays and vertical modes. These results have motivated theoretical and modeling studies of the waveguide behavior since it was first reported in 2010. [Work supported by ONR322OA.]

**11:10**

**2aUW8. Measured three-dimensional effects of mode-1 and mode-2 nonlinear internal waves on broadband acoustic wave propagation in shallow water.** Altan Turgut, Peter C. Mignerey, and Marshall H. Orr (Naval Res. Lab, Acoust. Div., Code 7161, Washington, DC 20375, altan.turgut@nrl.navy.mil)

Horizontal shadowing effects and frequency shifts of acoustic intensity level curves were measured with a bottomed horizontal array in the East China Sea during the summer of 2008. Low-frequency acoustic pulses were transmitted by two fixed sources at 33 km (270–330 Hz LFM) and 20 km (450–550 Hz LFM) range. Strong shadowing effects were observed when mode-1 nonlinear internal wave fronts were nearly parallel to the acoustic propagation path. Numerical studies indicated that shadowing effects are more complex for mode-2 nonlinear internal waves due to acoustic-mode dependent focusing and defocusing. These effects were further analyzed using 3-D PE simulations for more dynamic mode-2 nonlinear internal waves observed on the US New Jersey Shelf. The shadowing was less pronounced for internal waves with curved wave-fronts and small amplitudes. However, regular and irregular frequency shifts were still present. The experimental observations coupled with 3-D PE simulations suggest that nonlinear internal waves may be sensed and characterized via low-frequency acoustic signals. [Work supported by the Office of Naval Research.]

**11:25**

**2aUW9. Propagation of broad-band signals in shallow water in the presence of horizontal stratification.** Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mnt, Carmel, Haifa 31905, Israel, katz@phys.vsu.ru), Andrey Malykhin (Phys., Voronezh Univ., Voronezh, Russian Federation), and Alexander Tckhoidze (Marine GeoSci., Univ. of Haifa, Haifa, Israel)

Horizontal stratification in shelf zone of the ocean is provided by existence of coastal wedge, temperature fronts, nonlinear internal waves, slopes, and canyons, where typical scales are up to tenths of kilometer in range and up to tenths minutes in time, for some perturbations spatial scales are essentially different in different directions in horizontal plane. In this case, there is remarkable horizontal refraction in sound propagation and frequency dependence of horizontal ray trajectories. It means that Fourier components of wideband signal propagate along different paths joining source and receiver in the horizontal plane. Distribution of spectral components in horizontal plane has crescent-like shape and restricted by rays, corresponding to boundary frequencies in spectrum. Propagating wide-band signal has additional spectral distortion as a result of different phase shift for spectral components, propagating along different paths. Also there is difference in directions of wave vectors for difference spectral components (tangents to horizontal rays), leading to phenomena similar to spatial dispersion: different directions of phase and group velocities, compression and decompression of pulses, additional time delay of signal, etc. Mentioned phenomena are considered for models of coastal wedge and temperature fronts. Analytical estimations are presented, as well as results of numerical modeling.

TUESDAY MORNING, 3 DECEMBER 2013 UNION SQUARE 14, 9:00 A.M. TO 10:30 A.M.



## Meeting of the Standards Committee Plenary Group

to be held jointly with the meetings of the

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

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

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

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

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

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

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

P. J. Battenberg, U.S. Technical Advisor (TA) for IEC/TC 29, Electroacoustics
*3M Personal Safety Division, Detection Solutions, 1060 Corporate Center Drive Oconomowoc WI 53066*

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

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

| | | |
|---|---|---|
| Tuesday, 3 December 2013 | 10:45 a.m. - 12:00 noon | ASC S2, Mechanical Vibration & Shock |
| Tuesday, 3 December 2013 | 1:45 p.m. - 3:15 p.m. | ASC S1, Acoustics |
| Tuesday, 3 December 2013 | 3:30 p.m. - 5:00 p.m. | ASC S12, Noise |
| Wednesday, 4 December 2013 | 9:00 a.m. - 10:30 a.m. | ASC S3, Bioacoustics |
| Wednesday, 4 December 2013 | 10:45 a.m. - 12:00 noon | ASC S3/SC 1, Animal Bioacoustics |

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

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

| U.S. TAG Chair/Vice Chair | TC or SC | U.S. Parallel Committee |
|---|---|---|
| **ISO** | | |
| P.D. Schomer, Chair | **ISO/TC 43** Acoustics | ASC S1 and ASC S3 |
| P.D. Schomer, Chair | **ISO/TC 43/SC1** Noise | ASC S12 |
| M.A. Bahtiarian, Acting Chair | **ISO/TC 43/SC 3**, Underwater acoustics | ASC S1, ASC S3/SC 1 and ASC S12 |
| D.J. Evans, Chair | **ISO/TC 108** Mechanical vibration, shock and condition monitoring | ASC S2 |
| W.C. Foiles, Co-Chair | **ISO/TC 108/SC2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehiclesand structures | ASC S2 |
| D.J. Evans, Chair | **ISO/TC 108/SC3** Use and calibration of vibration and shock measuring instruments | ASC S2 |
| D.D. Reynolds, Chair | **ISO/TC 108/SC4** Human exposure to mechanical vibration and shock | ASC S3 |
| D.J. Vendittis, Chair | **ISO/TC 108/SC5** Condition monitoring and diagnostics of machine systems | ASC S2 |
| **IEC** | | |
| P.J. Battenberg, U.S. TA | **IEC/TC 29** Electroacoustics | ASC S1 and ASC S3 |

TUESDAY MORNING, 3 DECEMBER 2013 UNION SQUARE 14, 10:45 A.M. TO 12:00 NOON

## Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

A.T Herfat, Chair ASC S2
*Emerson Climate Technologies, Inc., 1675 West Campbell Road, PO Box 669, Sidney, OH 45365- 0669*

C.F. Gaumond, Vice Chair ASC S2
*Naval Research Laboratory, Code 7142, 4555 Overlook Ave. SW, Washington DC 20375-5320*

**Accredited Standards Committee S2 on Mechanical Vibration and Shock**. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

*People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock and condition monitoring, and four of its subcommittees, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.*

**Scope of S2**: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance and comfort.

## Session 2pAA

# Architectural Acoustics, Noise, and ASA Committee on Standards: Innovations and Challenges in Healthcare Acoustics

Erica Ryherd, Cochair
*Georgia Inst. of Technol., Mech. Eng., Atlanta, GA 30332-0405*

David M. Sykes, Cochair
*The Remington Group LP, 23 Buckingham St., Cambridge, MA 02138*

Kenric D. Van Wyk, Cochair
*Acoustics By Design, Inc., 124 Fulton St. East, Second Fl., Grand Rapids, MI 49503*

**Chair's Introduction—1:00**



### *Invited Papers*

**1:05**

**2pAA1. Research methods: Beta-testing Facilities Guideline Institute's acoustical criteria—A case study of continual improvement in healthcare facilities.** Kurt Rockstroh (Board of Directors, Facility Guidelines Inst., Falcon Pier, Boston, MA 55801, kurtr@steffian.com)

Continual improvement processes (CIP) present opportunities and challenges for researchers. In healthcare, FGI manages a four-year building-code-development cycle that is continually improved by research and sets standards that others follow (USGBC-LEED, ICC). Innovative, accelerated research methods are needed to continually improve healthcare environments during a period of rapid and profound change, so FGI's CIP method seeks to synchronize the work of code writers and independent researchers. One useful research method is to subject draft requirements to "beta-testing." FGI leaders (led by the presenter) did this to test new acoustical criteria on a 641,000 "hospital of the future" wing at Tufts University's Baystate Medical Center in Springfield MA (a major hospital and Level 1 trauma center). Planning began in 2006 (using draft #1 of the acoustical guidelines); occupancy began in late 2012 (after the guidelines' publication, Jan 2010). But only now, after a full year of occupancy, can the full cost and benefit of the improvements be measured using Federal HCAHPS (a new procedure required by the Affordable Care Act that impacts hospital reimbursements). "Beta test" results were factored into the forthcoming FGI 2014 edition. The "beta test" is the subject of independent, funded research currently underway.

**1:25**

**2pAA2. New healthcare safety risk assessment toolkit includes acoustics and noise control design measures for error prevention in medication safety zones.** Mandy Kachur (Soundscape Engineering LLC, 317 S Div. St. #170, Ann Arbor, MI 48104, mkachur@soundscapeengineering.com), Xiaobo Quan (The Ctr. for Health Design, Concord, CA), Daniel M. Horan (Cavanaugh Tocci Assoc., Inc., Sudbury, MA), and David M. Sykes (Acoust. Res. Council, Lincoln, MA)

The Facilities Guidelines Institute (FGI) manages the *Guidelines for Design and Construction of Hospitals and Outpatient Facilities*, which is used by the Joint Commission, many federal agencies, and authorities having jurisdiction in 42 states. The forthcoming 2014 edition calls for completion of a Safety Risk Assessment (SRA) to guide healthcare facility planning and design teams through systematic evaluation of safety issues in the built environment. In 2012, FGI and the federal Agency for Healthcare Research and Quality commissioned The Center for Health Design to oversee a three year effort to develop, test, and disseminate an SRA toolkit. The toolkit will consist of an online facility design questionnaire, supporting whitepapers and guidelines, and education for the healthcare community. The Acoustics Research Council was invited to contribute by integrating noise control language into the medication safety segment of the online questionnaire tool. The result is a question that specifically addresses acoustics and noise control design for error prevention in medication safety zones. In May 2013, the questionnaire content was finalized, and it will be validated, integrated, and evaluated during the subsequent two years. The co-authors are project participants.

**1:45**

**2pAA3. New healthcare acoustics subcommittee: Overview and call for participation.** Gary Madaras (Making Hospitals Quiet, Chicago Metallic, 4849 S. Austin Ave., Chicago, IL 60638, DoctorSonics@aol.com)

The Acoustical Society of America (ASA) Technical Committee on Architectural Acoustics (TCAA) formed a new Healthcare Acoustics subcommittee at the 165th meeting in Montreal. The 1st meeting of the Healthcare Acoustics subcommittee will be held at the 166th meeting in San Francisco. This presentation will provide an overview of the purpose of the subcommittee and introduce possible initiatives

on which the subcommittee may want to initially focus. *Examples*: 1. Despite financial incentive to make their hospitals quieter, most healthcare providers cannot significantly increase their "quiet-at-night" (HCAHPS) scores in their older or newer inpatient facilities. The national "quiet-at-night" (HCAHPS) score remains the lowest of all patient experience quality indicators. Changing things may require a significant paradigm shift in the architectural and acoustical communities. 2. The Facility Guidelines Institute (FGI) has developed its new *Guidelines for Design and Construction of Residential Health, Care, and Support Facilities* to be published in early 2014. Those Guidelines for eldercare facilities will include terms and concepts such as "nature sounds," "positive auditory distraction," "auditory landmark," "music," "quiet rooms," "quietly operating," etc., that have yet be fully defined acoustically. Please attend and participate.

**2:05**

**2pAA4. Value based education for healthcare design.** Edward Logsdon (D. L. Adams Associates, Inc., 1536 Ogden St., Denver, CO 80218, elogsdon@dlaa.com and Kenric Van Wyck (Acoustics By Design, Grand Rapids, MI 49503)

Accelerating the understanding and adoption of the FGI Guidelines is an essential part of the continuous improvement process—especially at a time when healthcare is coping with so much rapid change, and while the FGI Guidelines are becoming more accepted internationally. As part of the process to improve the acoustical environment in healthcare facilities, there is also the continual need to educate architects and designers concerning sound and vibration design criteria included in the FGI Guidelines. Designers need to understand how to apply the criteria, proposed changes to the guidelines, and how the changes will benefit the hospital by improving the patient experience. Using professional social networking, electronic, and face-to-face continuing education combined with research based design, professionals and the public can be educated on the merits of improvement in the acoustical design of healthcare facilities. Web-based education will attract a much wider audience worldwide and further international acceptance. Project case studies where FGI Guidelines were applied will identify the cost impact while increasing awareness and understanding.

**2:20**

**2pAA5. Top ten research needs in the decade ahead.** David M. Sykes and William J. Cavanaugh (Architectural Acoust., Rensselaer Polytechnic Inst., 31 Baker Farm Rd., Lincoln, MA 01773, david.sykes@remington-partners.com)

Independent, third-party research co-led by teams including medical personnel, engineers, and scientists is key to raising the bar on acoustical performance in healthcare facilities. While links between noise and health have been officially ignored in the United States for three decades, research over the past eight years in healthcare environments has significantly advanced understanding and created a small community of funding organizations and interested researchers. To make further progress, researchers must demonstrate clear links between noise/sound and patient outcomes as well as the performance effectiveness of healthcare professionals on critical factors such as error rates. Achieving useful, translational results requires: innovative research methods, collaborative funding mechanisms, and a clear focus on outcomes. With no Federal agency providing organized peer-review, it is essential for the acoustics research community to both agree on a research agenda and organize peer-review procedures that will enable continued progress among a widely distributed population of researchers working within the constraints of limited funding. The presenter worked with FGI to draft a "top 10" research agenda. He co-chairs the FGI Acoustics Working Group, edits the Springer Verlag "Quiet series," and has experimented with collaborative research funding mechanisms over the past three decades.

## *Contributed Paper*

**2:40**

**2pAA6. Acoustic quality of buildings: Contributions to the workers health of the health area.** Marta R. Macedo (Coordenação de Saúde do Trabalhador, Fundação Oswaldo Cruz, Av. Brasil 4365, Pavilhão Carlos Augusto da Silva, Sala 202, Rua Comandante Rubens Silva 90, ap. 203, bl 1, Rio de Janeiro, Rio de Janeiro 21040-360, Brazil, mribeiro@fiocruz.br), Marcia S. Almeida, Liliane R. Teixeira (Escola Nacional de Saúde Pública, Fundação Oswaldo Cruz, Rio de Janeiro, Rio de Janeiro, Brazil), Ana Paula Gama, Stephanie Livia S. Silva, Olga Dick (Coordenação de Saúde do Trabalhador, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil), Isabele C. Costa, Denise Torreao (Escola Nacional de Saúde Pública, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil), Paulo Roberto L. Jorge (Coordenação de Saúde do Trabalhador, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil), Paulo Marcelo S. Dias, Daniel Valente, and Diane R. Valente (Escola Nacional de Saúde Pública, Fundação Oswaldo Cruz, Rio de Janeiro, Brazil)

This work presents a case study carried through in a research center, education, assistance, and technological development in the areas of the health of the woman, child, and adolescent, situated in Rio de Janeiro, where pathologies of average and high complexity are taken care of. It intended detect environmental problems that could affect the health of workers of the institution, in order to support the elaboration of the action plan to mitigate urgent questions and to develop an architectonical project for a new center. The environments and processes of work had been evaluated, through a participative approach, being visited all the sectors of the center, carried through interviews half-structuralized with workers of all the sectors and promoted debates that had pointed the noise as one of the main factors of bother and stress. The observation in loco and analyze of the project allowed to detect that the architectural design and the disrespect of basic acoustics recommendations had contributed for this picture. Measurements carried through in some pointed areas as uncomfortable indicated to have exposition to sound pressure level above 75 dB(A). However, the resolution of the many of identified acoustic problems will have to wait the construction of the new headquarters.

**2:55–3:10 Break**

3:10

**2pAA7. Why alarm fatigue is a pivotal issue that affects the acoustical design of healthcare facilities.** Paul Barach MD (Res. Committee, Facility Guidelines Inst., 31 Baker Farm Rd., Lincoln, MA 01773, pbarach@gmail.com)

The U.S. FDA and Joint Commission designated "alarm fatigue" the "#1 priority in healthcare technology" in 2011–2012, acknowledging that this acoustical problem results in hundreds of patient deaths and thousands of injuries. The healthcare facilities industry has been slow to recognize that "alarm fatigue" is partly a facility design issue: i.e., a cacophony of recurrent noises from myriad uncorrelated medical devices, set at maximum loudness, occurring in hard-walled, reverberant spaces (such as patient rooms, ORs, and ICUs) produce elevated stress, sleep impairment, disorientation, and dangerously irrational, potentially deadly behavior. "Alarm fatigue" has been addressed as a human factors problem elsewhere: e.g., nuclear plant control rooms (after Three-Mile Island) and aircraft cockpits. In healthcare, it is imperative to engage architects, designers, acoustical engineers facility engineering staffs, and clinicians, who represent the "first line of defense" as the medical device industry requires 5–10 years to implement solutions. The presenter co-led a delegation of 12 distinguished members of the acoustics profession to the national summit on "alarm fatigue," Washington DC, 2011 and has co-authored peer-reviewed medical journal articles and a forthcoming FGI white paper on the subject. This presentation focuses on solutions, challenges, and the research roadmap needed to address "alarm fatigue."

3:30

**2pAA8. The healthcare acoustics research team: Bridging the gap between architecture, engineering, and medicine.** Erica Ryherd (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Mech. Eng., Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu), Jeremy Ackerman (Dept. of Emergency Medicine, Emory Univ., Atlanta, GA), Craig Zimring (College of Architecture, Georgia Inst. of Technol., Atlanta, GA), and Kerstin Persson Waye (Occupational and Environ. Medicine, Gothenburg Univ., Gothenburg, Sweden)



Healthy soundscapes are paramount to the missions of hospitals: patients need to sleep and heal without environmental stressors; staff, patients, and family need to communicate accurately but privately; staff need to be able to localize alarms and calls for help. This talk discusses recent findings from the Healthcare Acoustics Research Team (HART), an international, interdisciplinary collaboration of specialists in architecture, engineering, medicine, nursing, and psychology. Members of the HART network are actively engaged in research in the United States and Sweden, having worked in more than a dozen hospitals and a broad range of unit types including adult and neonatal intensive care, emergency, operating, outpatient, long-term care, mother-baby, and others. Highlights will include projects relating noise and room acoustic measures to staff and patient response in addition to studies evaluating impacts of acoustic retrofits. Results show that effective hospital soundscapes require a complex choreography of architectural layout, acoustic design, and administrative processes that is only beginning to be fully understood.

3:50

**2pAA9. Acoustic comfort in healthcare facilities—What is it? and What does it mean to both patients and medical staff?** Kenneth P. Roy (Bldg. Products Technol. Lab, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrong.com)

Acoustic comfort is all about meeting the acoustic needs of the patient, family, professional staff in hospitals, clinics, pharmacies, etc. These needs include low distraction and annoyance for patients to fully enjoy a "healing environment", high levels of speech intelligibility to support communications and to help reduce medical errors, and adequate speech privacy to meet HIPAA and other privacy needs. How to design for these, and how to verify performance—these are the key issue to be addressed. Architectural choices need to be made with a view for acoustic comfort in addition to all the other relevant design factors. Facility evaluations as part of the commissioning process need to include both objective and subjective surveys to push forward evidence based design approaches for others to take advantage of in future designs. All of these issues are part of the ongoing discussions and field studies that we have been having, and will continue to have.

4:10

**2pAA10. Analysis of granular hospital sound level measurements.** Benjamin Davenny (Acentech Inc., Cambridge, MA), Aaron Betit (Acentech Inc., 601 South Figueroa St., Ste. 4050, Los Angeles, CA 90017, abetit@acentech.com), and William Yoder (Acentech Inc., Cambridge, MA)

Remote sound monitoring systems were deployed in several locations in a hospital to measure sound levels due to building system and activity noise. In addition to 5 min interval data, granular sound levels were recorded several times per second to provide flexibility in data analysis. Analysis of this information will be presented.

4:30

**2pAA11. Estimation of noise induced hearing loss because of indoor and outdoor environment noise factors in Turkish healthcare facilities: A survey of hospitals in Turkey.** Filiz Kocyigit (Architecture, Atilim Univ., İncek, Ankara, Turkey, filizbk@gmail.com)

This article aims to evaluate effect of indoor and outdoor environment noise factors in healthcare facilities which affect the work area due to noise induced hearing loss, and also to determine their relationship with the architectural design of the building. As a case study; noise levels, in five state and private healthcare centers, including medical schools, research hospitals, and state hospitals in Turkey were measured. They were compared with similar healthcare centers in the United States. Results include equivalent sound pressure levels (Leq for 5 min from 20 spots in each area), and Lmax–Lmin evaluated as a function of location, frequency, time, and days of the week. Research showed that no location was in compliance with current World Health Organization Guidelines, and a review of

objective data indicated that this was true of hospitals throughout the world. Noise induced hearing loss on continuously users had been estimated with these results in selected hospitals. Data gathered at various hospitals for last decay indicate a trend of increasing noise levels during daytime and nighttime hours. The implications of these results were significant for patients, visitors, and hospital staff.

**4:50**

**2pAA12. Renovation of neonatal intensive care unit per the Facilities Guideline Institute's 2010 Acoustical Design Guidelines.** Erik Miller-Klein (SSA Acoust., LLP, 222 Etruria St, Ste. 100, Seattle, WA 98109, erik@ssaacoustics.com)

We did an evaluation of the acoustical conditions of the existing neonatal intensive care unit at the St. Joseph's Hospital in Tacoma, Washington, and issued detailed renovation recommendations that were integrated into the final design. At the conclusion of construction, our team was able to evaluate the improvements of the acoustical conditions per the 2010 FGI Guidelines and the original performance. These included noise impacts from door closures, curtains, alarms, and the conversations.

**5:10–6:10 Panel Discussion**

TUESDAY AFTERNOON, 3 DECEMBER 2013 UNION SQUARE 23/24, 1:00 P.M. TO 3:20 P.M.

## Session 2pABa

## Animal Bioacoustics and Acoustical Oceanography: Broadening Applications of Tags to Study Animal Bioacoustics II

Marla M. Holt, Cochair
*NOAA NMFS NWFSC, 2725 Montlake Blvd. East, Seattle, WA 98112*

Alison K. Stimpert, Cochair
*Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA 93943*

### *Invited Papers*

**1:00**

**2pABa1. Conservation applications of baseline acoustic tag data from right whales.** Susan Parks (Dept. of Biology, Syracuse Univ., 114 Life Sci. Complex, Syracuse, NY 13244, sparks@syr.edu), Douglas P. Nowacek (Nicholas School of the Environ. and Pratt School of Eng., Duke Univ. Marine Lab., Beaufort, NC), Mark Johnson, and Peter L. Tyack (Sea Mammal Res. Unit, Scottish Oceans Inst., School of Biology, Univ. of St. Andrews, St. Andrews, Scotland, United Kingdom)

Data collected from archival acoustic tags have dramatically improved our understanding of marine mammal subsurface behavior. Some of the earliest tag deployments were made on the endangered North Atlantic right whale (*Eubalaena glacialis*) to better understand their behavior and to contribute to their conservation. Over the past 15 years, the database of acoustic tag records collected from a diverse cross section of the population has served as a valuable resource for research. Acoustic tag data have provided critical behavioral information including details on individual calling behavior in designated critical habitat feeding grounds, behavioral responses to controlled exposures of acoustic signals including conspecific and man-made signals, acoustic responses to vessel noise, and ground truth data for passive acoustic monitoring efforts. Current studies continue to utilize tag data to shed light on the individual, seasonal, and regional variation in acoustic behavior of right whales, particularly focused on the shallow water coastal habitats of the calving grounds off of Florida and Georgia and in the migration corridor off the Eastern United States. These studies have contributed data that are crucial for targeted conservation efforts and highlight the value of long-term databases of tag data in baleen whale research.

**1:20**

**2pABa2. Controlled sound exposure experiments to measure marine mammal reactions to sound: Southern California behavioral response study.** Brandon L. Southall (SEA, Inc., 9099 Soquel Dr., Ste. 8, Aptos, CA 95003, Brandon.Southall@sea-inc.net), John Calambokidis (Cascadia Res. Collective, Olympia, WA), Moretti David (Naval Undersea Warfare Ctr., Newport, RI), Jay Barlow (Southwest Fisheries Sci. Ctr., La Jolla, CA), Stacy DeRuiter (CREEM The Observatory, St. Andrews, Scotland, United Kingdom), Jeremy Goldbogen (Cascadia Res. Collective, Olympia, WA), Ari Friedlaender (Long Marine Lab., Univ. of California, Santa Cruz, Santa Cruz, CA), Elliott Hazen (NOAA-PFEL, Pacific Grove, CA), Alison Stimpert (Naval Postgrad. School, Monterey, CA), Arranz Patricia (Sea Mammal Res. Unit, Scottish Oceans Inst., St. Andrews, Scotland, United Kingdom), Erin Falcone, Greg Schorr, Annie Douglass (Cascadia Res. Collective, Olympia, WA), Chris Kyburg (SPAWAR Systems Ctr., San Diego, CA), and Peter Tyack (Sea Mammal Res. Unit, Scottish Oceans Inst., St. Andrews, Scotland, United Kingdom)

SOCAL-BRS is an inter-disciplinary collaboration designed to increase understanding of marine mammal behavior and provide a robust scientific basis for estimating risk and minimizing effects of mid-frequency military sonar systems. Data were collected using visual observations, passive acoustic monitoring, animal-attached acoustic and movement tags, photo ID, biopsy, and controlled sound exposure experiments on over 20 cetacean species in biologically important areas throughout the southern California Bight. Ninety-six individuals of ten species were tagged with six tag types, including two species [Baird's beaked whale (*Berardius bairdii*), Risso's dolphin (*Grampus griseus*)] not previously studied with such tools. Fifty-six controlled CEEs were conducted using protocols and protective measures to ensure animals were not harmed. Simulated sonar signals were projected through a deployed sound source and changes in vocal, diving, and horizontal movement behavior were measured. Results demonstrate that Cuvier's beaked whales (*Ziphius cavirostris*) react most strongly to simulated sonar exposures with clear changes in vocal and diving behavior and avoidance responses at low received sound levels. Blue whale (*Balaenoptera musculus*) responses are more variable, depending on complex interactions of exposure and behavioral conditions. Ongoing efforts include expanding sample sizes in other species using simulated sounds and the novel inclusion of operational mid-frequency sonars.



**1:40**

**2pABa3. Noise design tradeoffs for a general-purpose broadband acoustic recording tag.** William C. Burgess (Greeneridge Sci., Inc., 6060 Graham Hill Rd. Stop F, Felton, CA 95018, burgess@greeneridge.com), Susanna B. Blackwell (Greeneridge Science, Inc., Aptos, CA), and Patrick Dexter (Greeneridge Science, Inc., Ojai, CA)

To design instruments for general-purpose use invokes both blessings and curses; blessings because a flexible product may leverage its development effort across many different applications, but curses because it may not be perfect for any of them. Designing for noise performance of broadband acoustic recording tags epitomizes this tradeoff. Lower self noise typically requires more power and a larger transducer, and may come at the expense of clipping strong sounds, all of which impact tag applications. Higher self noise, however, reduces detection range under quiet conditions and diminishes utility for noise monitoring. The electronic design of the Acousonde™ acoustic/ultrasonic recording tag navigates this challenge using two acoustic channels with very different gains (29–49 versus 14–34 dB), noise (minimum 40 versus 70 dB re 1 $\mu Pa^2/Hz$ at 1 kHz), and bandwidths (42 versus 9.3 kHz). Mechanically, design for hydrodynamics reduces turbulent flow noise especially once the tag aligns with flow direction due to self-orienting. As a result of these design elements, the Acousonde's noise performance is comparable to that of a much larger recording instrument while preserving capability as an animal tag. [Development supported by ONR.]

**2:00**

**2pABa4. The next generation of multi-sensor acoustic tags: Sensors, applications, and attachments.** Douglas Nowacek (Nicholas School of the Environ. and Pratt School of Eng., Duke Univ. Marine Lab., 135 Duke Marine Lab Rd., BEAUFORT, NC 28516, dpn3@duke.edu), Matthew Bowers (Marine Sci. and Conservation, Duke Univ. Marine Lab, BEAUFORT, NC), Andrew Cannon (1900 Eng., Greenville, SC), Mark Hindell (Inst. of Marine and Antarctic Sci., Univ. of Tasmania, Hobart, TAS, Australia), Laurens E. Howle (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), Mark M. Murray (Mech. Eng., US Naval Acad., Annapolis, MD), Dan Rittschof (Marine Sci. and Conservation, Duke Univ. Marine Lab, Beaufort, NC), K. Alex Shorter, and Michael Moore (Biology Dept., Woods Hole Oceanographic Inst, Woods Hole, MA)

From Kooyman's 1963 wind-up kitchen timer TDR, multi-sensor tags have evolved significantly over the last twenty years. These advancements, including high fidelity acoustics, have been driven by improved sensing and electronics technology, and resulted in highly integrated mechatronics systems for the study of free ranging animals. In the next decade, these tags will continue to improve, and promising work has begun in three key areas: (i) new sensors; (ii) expanding uses of existing sensors; and (iii) increasing attachment duration and reliability. The addition of rapid acquisition GPS and the inclusion of gyroscopes to separate the dynamic acceleration of the animal from gravitational acceleration, are underway but not widely available to the community. Existing sensors could be used for more and different applications, e.g., measuring ambient ocean noise. Tags attached to pinnipeds in the Southern Ocean, for example, could provide noise measurements from remote areas. Finally, attachment duration has been limiting for cetaceans because the suction cups typically used do not reliably stay attached for more than a day. We will present data on engineering efforts to improve attachments: (i) improved tag hydrodynamics; (ii) incorporating bio-compatible glues; and (iii) micro structuring tag components to utilize hydrostatic forces and enhance adhesion.

2:20

**2pABa5. The relationship between vessel traffic and noise levels received by killer whales.** Juliana Houghton (School of Aquatic & Fishery Sci., Univ. of Washington, Box 355020, Seattle, WA 98195, stephj5@uw.edu), Marla Holt (NOAA/NMFS/Northwest Fisheries Sci. Ctr., Seattle, WA), Deborah Giles (Dept. of Wildlife, Fish, and Conservation Biology, Univ. of California, Davis, CA), Candice Emmons, Brad Hanson (NOAA/NMFS/Northwest Fisheries Sci. Ctr., Seattle, WA), Jeff Hogan (Cascadia Res. Collective, Olympia, WA), Trevor Branch, and Glenn VanBlaricom (School of Aquatic & Fishery Sci., Univ. of Washington, Seattle, WA)

Cetaceans that rely on their acoustic environment for key life history strategies are susceptible to noise effects from anthropogenic use such as ecotourism. Endangered Southern Resident killer whales (SRKW) are the primary target for vessel-based whale-watching in the Salish Sea. Vessel interactions and associated noise have been identified as potential stressors for SRKW. Previous research has indicated that both stressors negatively impact SRKW; however, there is a missing link between vessel characteristics/behavior and noise levels actually received by individual whales. To investigate this relationship, data were collected concurrently using mobile remote sensing survey equipment packages and digital acoustic recording tags. This allowed us to obtain precise geo-referenced vessel data and noise levels received by the whales. We used linear regression to summarize patterns in vessel characteristics and relate them to received noise levels. Received noise levels (RNL) were correlated with the number of vessels. RNL also increased when larger vessels were present or when vessels were traveling at relatively high speed. These findings facilitate improved understanding of the contributions of vessel characteristics to the noise levels received by individual cetaceans. Results from this study can be used to refine existing vessel regulations in order to better manage SRKW to recovery.

2:35

**2pABa6. Use of an animal-borne active acoustic tag to conduct minimally-invasive behavioral response studies.** Selene Fregosi, Holger Klinck (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Hatfield Marine Sci. Ctr., 2030 SE Marine Sci. Dr., Newport, OR 97365, selene.fregosi@noaa.gov), Markus Horning (Marine Mammal Inst., Oregon State Univ., Newport, OR), David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), Daniel P. Costa (Dept. of Ecology and Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA), David A. Mann (Loggerhead Instruments, Sarasota, FL), Kenneth Sexton (The Sexton Co., Salem, OR), and Luis Huckstadt (Dept. of Ecology and Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA)

A pilot study was conducted to evaluate the potential of animal-borne active and passive acoustic tags for conducting minimally-invasive behavioral response studies on pinnipeds. A prototype tag was developed and tested on juvenile northern elephant seals (*Mirounga angustirostris*) using translocation experiments at Año Nuevo State Park, CA, USA, in spring 2012. The principal scientific questions of this pilot study were (1) do low-intensity sounds emitted by an animal-borne tag elicit behavioral responses, and (2) are potential animal responses related to signal content (e.g., threatening vs non-threatening)? Preliminary results indicate that (1) low-intensity sounds emitted by animal-borne tags elicit distinct behavioral responses, (2) these responses appear related to signal content, and (3) the responses may differ based on depth, bathymetry, and location. The results of the study show the promise of this approach as a minimally invasive and cost-effective method to investigate animal responses to underwater sounds, as well as a method to develop mitigation strategies. We are currently in the process of improving the tag design for future field efforts with the goal to increase the sample size, range of acoustic stimuli, and age/sex classes of tagged seals. [Funding from NOAA/NMFS Ocean Acoustics Program.]

2:50

**2pABa7. Putting tags in the researcher's toolkit: An examination of the strengths, limitations, and added-value from animal tagging.** Robert Gisiner (NAVFAC EXWC EV, US Navy, 1000 23rd Ave., Port Hueneme, CA 93043, bob.gisiner@navy.mil)

The US Navy, through the Office of Naval Research and other offices, has focused on improving animal tags, reducing cost, and increasing availability. From data-rich packages like video and acoustic dataloggers to simple location-only tags, tags provide a variety of new data to studies of marine animals and their ecosystems. Tags realize their full potential when calibrated or validated against other existing alternative sensor systems like visual surveys, photo-identification, genetics, and acoustic monitoring. When tag cost, cost of delivery and recovery or monitoring are weighed against the data uniquely available from tags, an integrated data collection strategy involving animal tagging can be developed to generate the best data at the optimal total cost for a given research or resource management scenario.

**3:05–3:20 Panel Discussion**

# Session 2pABb

## Animal Bioacoustics: Noise Impacts on Marine Life

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

### *Contributed Papers*

3:30

**2pABb1. Passive acoustic monitoring for marine mammals during Navy explosives training events off the coast of Virginia Beach, Virginia.** Cara F. Hotchkin, Mandy Shoemaker, Anurag Kumar (Naval Facilities Eng. Command, Atlantic, 6506 Hampton Blvd., Norfolk, VA 23508, cara.hotchkin@navy.mil), Carl Hager (U.S. Naval Acad., Annapolis, MD), David MacDuffee, Jene Nissen, and Ronald Filipowicz (U.S. Fleet Forces Command, Norfolk, VA)

Navy training events involving the use of explosives pose a potential threat to marine mammals. This study used passive acoustic and visual monitoring data to evaluate marine mammals' behavioral responses to noise from explosive events. Monitoring was conducted during five training events in the Virginia Capes (VACAPES) Range Complex during August/September of 2009–2012. Passive acoustic monitoring methods ranged from a single hydrophone to an array of sonobuoys monitored in real time. Visual monitoring effort over the five events totaled approximately 34 h (day before events: 10.1 h; days of events: 22.3 h; day after events: 1.5 h), yielding a total of 27 marine mammal sightings. Approximately 54 h of acoustic data were collected before, during, and after the five events. Behavioral changes were evaluated based on analysis of vocalizations detected before, during, and after explosions and concurrent data from visual sightings. For time periods with both visual and acoustic monitoring data, detection methods were compared to evaluate effectiveness. Continuing use and evaluation of both visual and passive acoustic methods for monitoring of explosive training events will improve our knowledge of potential impact resulting from explosive events and help improve management and conservation of marine mammals.

3:45

**2pABb2. Use of Automated passive acoustic monitoring methods for monitoring for marine mammals in conjunction with US Navy Mid-frequency Active Sonar training events.** Stephen W. Martin, Roanne A. Manzano-Roth, and Brian M. Matsuyama (SSC PAC, 53560 Hull St., Code 71510, San Diego, CA 92152, steve.w.martin@navy.mil)

Automated passive acoustic detection, classification, and localization (DCL) methods are employed to deal with large volumes of acoustic data to support estimating the sound pressure levels (SPLs) that marine mammals are exposed to from mid-frequency active sonar (MFAS) during US Naval training events. These methods are applied to a training event involving MFAS conducted February 2012 in Hawaiian waters with thirty one hydrophones of data collected continuously over an 11 day period. The automated methods detect and determine locations of marine mammals, specifically minke and beaked whales, and the times of the MFAS transmissions utilizing custom C++ algorithms. Streamlined manual validation methods are employed which utilize custom MATLAB display routines. Animal locations uncertainties are addressed for the two different species. Once the transmitting ship and animal locations are determined acoustic propagation modeling is utilized to estimate the sound pressure levels (in dB re 1 micro Pascal) that an animal, or group of animals, were exposed to. Surface ducted propagation conditions can result in species such as beaked whales being exposed to over 30 dB higher SPL's when they return to the surface to breathe compared to when at depth foraging.

4:00

**2pABb3. Impact of underwater explosions on cetaceans.** Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Amanda J. Debich, Ana Širović (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), James V. Carretta (Southwest Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, San Diego, CA), Jennifer S. Trickey, Rohen Gresalfi (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA), Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA), Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, San Diego, CA)

Use of seal bombs to deter sea lions from being caught in nets and preying on catch is a common practice for a number of fisheries. Purse seine fisheries in Southern California target primarily squid, but also scombrids and baitfish such as sardine and anchovy, while set gillnet fisheries' primary catch are halibut and white seabass. All of these fisheries use seal bombs as deterrents. Continuous passive acoustic recordings at several sites in the Southern California Bight collected since 2007 revealed an extensive use of smaller explosives, most likely seal bombs, during nighttime hours with a seasonal occurrence matching fishery activities. During several months of the year they were used all night, every night. The median occurrence of explosions when detected was 8 per hour; however, during periods of high fishing effort they reached up to 480 explosions per hour. From behavioral response and opportunistic studies we know that beaked whales as well as endangered blue whales react negatively to anthropogenic sound sources. We are testing the hypothesis that these underwater explosions have a suppressive effect on the acoustic behavior and therefore the communication and foraging of cetaceans, possibly leading to impacts on the individual fitness and overall population health.

4:15

**2pABb4. Monitoring of marine mammal occurrence and acoustic behaviors in relation to mid-frequency active sonar using autonomous recorders deployed off the undersea warfare training range, Florida.** Thomas F. Norris, Julie Oswald, Tina M. Yack, Elizabeth Ferguson (Bio-Waves, Inc., 144 W. D St., Ste. #205, Encinitas, CA 92024, thomas.f.norris@bio-waves.net), Anurag Kumar (Naval Facilities Eng. Command Atlantic, U.S. Navy, Norfolk, VA), Jene Nissen (U.S. Fleet Forces Command, U.S. Navy, Norfolk, VA), and Joel Bell (Naval Facilities Eng. Command Atlantic, U.S. Navy, Norfolk, VA)

Passive acoustic data were collected from nine Marine Autonomous Recording Units (MARUs) deployed 60–150 km in an area that coincides with the U.S. Navy's planned Undersea Warfare Training Range (USWTR) off Jacksonville FL. MARUs were deployed for 26 days during fall 2009, and 37 days in winter 2009–2010. Data were manually reviewed for marine mammal vocalization events, man-made noise, and mid-frequency active sonar events, which were logged using TRITON software. Seasonal and diel patterns were characterized qualitatively. Patterns and probabilities of vocalization events by species, or species groups, were related to sonar events. Vocalizations were detected for minke whales, North Atlantic right

2p TUE. PM

whales, sei whales, humpback whales, sperm whales, the blackfish group, and delphinids. Minke whale pulse-trains occurred almost continuously during the winter deployment but were absent in fall. Right whale events occurred mostly during winter at shallow-water sites, but unexpectedly were also detected at deep-water sites. Sperm whale events occurred exclusively near the continental shelf break and exhibited a strong diel pattern. Minke whale events had a strong negative relationship with sonar events. These results provide an initial assessment of marine mammal occurrence within the Navy's planned USWTR, and provide new information on vocalization events in relation to sonar.

**4:30**

**2pABb5. Vocalization behaviors of minke whales in relation to sonar in the planned Undersea Warfare Training Range off Jacksonville, Florida.** Talia Dominello, Thomas Norris, Tina Yack, Elizabeth Ferguson, Cory Hom-Weaver (Bio-Waves Inc., 364 2nd St. #3, Encinitas, CA 92024, talia.dominello@bio-waves.net), Anurag Kumar (Naval Facilities Eng. Command Atlantic, Norfolk, VA), Jene Nissen (U.S. Fleet Forces Command, Norfolk, VA), and Joel Bell (Naval Facilities Eng. Command Atlantic, Norfolk, VA)

Nine Marine Autonomous Recording Units (MARU's) were deployed in a rectangular array at a site coinciding with the United States (U.S.) Navy's planned Undersea Warfare Training Range (USWTR) approximately 60–150 km offshore Jacksonville, FL (13 September to 8 October and 3 December to 8 January, 2009–2010) at shallow, mid-depth, and deep sites (45, 183, 305 m). Data were reviewed in detail using TRITON (Wiggins, 2007). Event logs were created for each day at every site. Custom-written MATLAB scripts were used to calculate the probability of minke whale vocalization events occurring in the presence and in the absence of mid-frequency sonar. Minke whale vocalization events were completely absent in the fall deployment period, but occurred almost continuously during the winter deployment, indicating a strong seasonal pattern of occurrence. Minke whale vocalizations were detected most frequently at deep-water sites, and only at low levels (<0.03% of time) at shallow-water sites. Results of the probability analysis indicated a strong negative correlation to sonar. Minke whale vocalization events were greatly reduced, or completely ceased, during most days with nearly continuous sonar events during an approximate 3-day period. To our knowledge, such changes in acoustic behaviors of minke whales in relation to sonar have not been reported before.

**4:45**

**2pABb6. Behavioral responses of California sea lions (*Zalophus californianus*) to controlled exposures of mid-frequency sonar signals.** Dorian S. Houser (Dept. of Conservation and Biological Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), Stephen W. Martin, and James J. Finneran (US Navy Marine Mammal Program, SSC Pacific, San Diego, CA)

Acoustic dose-response functions can be used to explore the relationship between anthropogenic noise exposure and changes in marine mammal behavior. Fifteen sea lions participated in a controlled exposure study to determine the relationship between the received sound pressure level (SPL) of a mid-frequency sonar signal (1-s duration, 3250-3450 Hz) and behavioral deviations from a trained behavior. Sea lions performed 10 control trials followed by 10 exposure trials within an open-water enclosure. Acoustic playbacks occurred once during each exposure trial when the sea lion crossed the middle of the enclosure. Received levels, ranging from 125 to 185 dB re 1 $\mu$Pa (rms) SPL, were randomly assigned but were consistent across all trials for each individual. Blind scoring of behavioral responses was performed for all trials. A canonical correlation analysis indicated that cessation of the trained behavior, haul-out, a change in respiration rate, and prolonged submergence were reliable response indicators. Sea lions showed both an increased responsiveness and severity of response with increasing received SPL. No habituation to repeated exposures was observed, but age was a significant factor affecting the dose-response relationship. Response patterns and factors affecting behavioral responses were different from those observed in bottlenose dolphins and are indicative of species-specific sensitivities.

**5:00**

**2pABb7. The characteristics of boat noise in marine mammal habitats.** Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

Environmental management of underwater noise tends to focus on sources related to offshore oil and gas exploration and navy operations. However, in many, specifically near-shore habitats, boat noise prevails, yet remains largely unregulated. Underwater noise of personal watercraft (jet skis) was recorded in Queensland, and consisted of broadband energy between 100 Hz and 10 kHz attributed to the vibrating bubble cloud generated by the jet stream, overlain with frequency-modulated tonals corresponding to impeller blade rates and harmonics. Broadband monopole source levels were 149, 137, and 122dB re 1 $\mu$Pa @ 1m (5th, 50th, and 95th percentiles). Underwater noise of zodiacs (inflatable boats with outboard motors) operated by whale-watching companies was recorded in southern British Columbia. At slow cruising speeds (10 km/h), underwater noise peaked between 50 and 300 Hz; at fast traveling speeds (55 km/h), underwater noise peaked between 100 and 3000 Hz, exhibiting strong propeller blade rate tonals at all speeds. Broadband source levels increased with speed from 126 to 170dB re 1 $\mu$Pa @ 1m according to SL = 107 + 32 log10(-speed/km/h). Even though noise levels from jetskis are lower than those of propeller-driven boats, it is not necessarily the broadband source level that correlates with the bioacoustic impact on marine fauna.

**5:15**

**2pABb8. Complex masking scenarios in Arctic environments.** Jillian Sills (Ocean Sci., Univ. of California at Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95050, jmsills@ucsc.edu), Colleen Reichmuth (Inst. of Marine Sci., Univ. of California at Santa Cruz, Santa Cruz, CA), and Brandon L. Southall (Southall Environmental Associates, Aptos, CA)

Critical ratios obtained using octave-band noise and narrowband signals provide a useful first approximation for understanding the effects of noise on hearing. When considering realistic listening scenarios, it may be necessary to examine the effects of spectrally complex, time-varying noise sources on an animal's ability to detect relevant signals. In the case of Arctic seals, the increasing prevalence of seismic exploration makes an examination of masking by impulsive sounds particularly relevant. However, the characteristics of received sounds from airgun operations vary dramatically depending on the seismic source, environmental parameters, and distance. In order to determine the potential for auditory masking by airguns, we developed a paradigm to quantify the influence of spectral and temporal variations in typical seismic noise on signal detectability. This method calls for calculation of detection probabilities for seals listening for the same signal embedded at different time windows within a background of distant airgun noise. We believe this approach will enable an experimental assessment of masking potential by impulsive noise as distance between the receiver and source is increased. Such an assessment will aid in determining the extent to which standard critical ratio data can be reasonably applied in complex masking scenarios. [Work supported by OGP-JIP.]

## Session 2pBA

## Biomedical Acoustics: Nanobubbles, Nanoparticles, and Nanodroplets for Biomedical Acoustics Applications

Jonathan Mamou, Cochair
*F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038*

Jeffrey A. Ketterling, Cochair
*Riverside Res., 156 William St., New York, NY 10038*

### *Invited Papers*



**1:15**

**2pBA1. Ultrasound-targeted delivery of systemically administered therapeutic nanoparticles.** Kelsie Timbie, Caitlin Burke (Biomedical Eng., Univ. of Virginia, Box 800759, Health System, Charlottesville, VA 22908, rprice@virginia.edu), Elizabeth Nance, Graeme Woodworth (Ctr. for Nanomedicine, Johns Hopkins Univ., Baltimore, MD), Grady W. Miller (Biomedical Eng., Univ. of Virginia, Charlottesville, VA), Justin Hanes (Ctr. for Nanomedicine, Johns Hopkins Univ., Baltimore, MD), and Richard J. Price (Biomedical Eng., Univ. of Virginia, Charlottesville, VA)

The ultrasound (US)-targeted delivery of systemically administered drug and gene-bearing nanoparticles has emerged to become a robust area of investigation with clear clinical potential. Such approaches typically entail the concurrent injection of contrast agent microbubbles (MBs) and nanoparticles, followed by the application of US to the region of interest. US-activated MBs disrupt the surrounding microvessel, permitting nanoparticle delivery with precise spatial localization. Our group has previously shown that US-targeted nanoparticle delivery can amplify collateral artery growth, that the binding of nanoparticles to MBs enhances nanoparticle delivery, that non-viral gene nanocarrier transfection is dependent on both MB diameter and US pressure, and that solid tumor growth can be controlled by the US-targeted delivery of 5 FU nanoparticles. More recent studies center on developing MRI-guided focused ultrasound (FUS) for nanoparticle delivery across the blood brain-barrier (BBB), which is the foremost impediment to drug treatment for most central nervous system diseases. Importantly, densely PEGylated nanoparticles that have been specifically designed to penetrate brain tissue (i.e., brain-penetrating nanoparticles, BPNs) are capable of crossing the BBB after it is opened with FUS and MBs, thereby supporting the use of gene- and drug-bearing BPNs in combination with MR-guided FUS in treating disorders and pathologies of the CNS.

**1:35**

**2pBA2. Ultrasound-mediated stimulation of nanoparticles for therapeutic applications.** Tyrone M. Porter (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, tmp@bu.edu)

Ultrasound has the unique capability to change the pressure and temperature of tissue in a noninvasive and highly localized manner. Consequently, ultrasound is an attractive tool for enhancing the delivery of drugs to targeted cells and tissues. This talk will review a variety of stimuli-responsive drug carriers (i.e., liposomes, nanoparticles, emulsions, etc.) and the role of ultrasound in facilitating transport and drug delivery.

**1:55**

**2pBA3. Low boiling point perfluorocarbon nanodroplets for biomedical imaging and therapeutics.** Paul S. Sheeran (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ., Chapel Hill, NC), Terry O. Matsunaga (Radiology, Univ. of Arizona, Tucson, AZ), and Paul A. Dayton (Biomedical Eng., Univ. of North Carolina at Chapel Hill and North Carolina State Univ.,Chapel Hill, NC 27599, padayton@bme.unc.edu)

Perfluorocarbon (PFC) droplets have been examined for biomedical applications for more than a decade. Of particular interest are "phase change" PFC droplets, which can convert from liquid to gaseous phase with an acoustic energy input. In liquid form, these droplets demonstrate stability not capable with microbubbles, but in gaseous form the resultant bubble provides the cavitation source that is crucial for ultrasound imaging or therapeutic mediation. Traditionally, phase-change droplets have utilized PFCs with boiling points close to body temperature, so that they could be readily activated with acoustic energy input. However, the desire reduce these droplets to hundred-nanometer sizes in order to reach the extravascular space has made PFC selection more complicated. Specifically, the influence of surface tension on droplets that are significantly smaller than a micron elevates the boiling point past that of the bulk fluid, making hundred-nanometer sized droplets challengingly difficult to vaporize with diagnostic ultrasound energy levels. Our solution has been a unique approach to condense low boiling point gaseous perfluorocarbons, creating a population of meta-stable PFC nanodroplets. The resulting nanodroplets exhibit stability, yet have very low energy activation requirements. We discuss advantages and disadvantages of this new formulation, and demonstrate applications in biomedical imaging and therapeutics.

2:15

**2pBA4. Entropy imaging and nanostructures.** Michael Hughes (School of Medicine, Washington Univ., 1632 Ridge Bend Dr., St Louis, MO 63108, mshatctrain@gmail.com), John E. McCarthy (Dept. of Mathematics, Washington Univ., St Louis, MO), Jon N. Marsh, and Samuel A. Wickline (School of Medicine, Washington Univ., St Louis, MO)

Virtually all imaging devices today function by collecting either electromagnetic or acoustic waves and using the energy carried by these waves to determine pixel values to build up what is basically an "energy" picture. However, waves also carry information and this can also be used to determine the pixel values in an image. We have employed several measures of information all of which are based on different forms of entropy. Numerous published studies have demonstrated the advantages of entropy, or "information," imaging over conventional methods in materials characterization and medical imaging. Moreover, the technique is robust as these results were obtained using a variety of imaging systems having different frequency ranges and transducers. Similar results have also been obtained using microwaves. We will present the results of several entropy-imaging-based *in vivo* studies. The first of these were based on a non-gaseous liquid-filled nanoparticle contrast agent used to image tumors. We will also present a study of therapy monitoring of muscular dystrophy conducted without contrast. In both studies entropy images were able to detect changes in backscattered ultrasonic signals that were not detectable using conventional techniques.

2:35

**2pBA5. Characterization of liquid-filled nanoparticles for detection and drug-delivery in tumors.** Nicolas Taulier, Thomas Payen, Sara Jafari (Parametric Imaging Lab, CNRS - Univ. Pierre and Marie Curie, 15 rue de l'ecole de medecine, Paris 75006, France, lori.bridal@upmc.fr), Jonathan Mamou (F.L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), Fattal Elias, Nicolas Tsapis (Institut Galien Paris-Sud, UMR 8612, LabEx LERMIT, Univ Paris-Sud and CNRS, Châtenay-Malabry, France), and Lori Bridal (Parametric Imaging Lab, CNRS - Univ. Pierre and Marie Curie, Paris, France)

Liquid-filled nanoparticles provide long half-life circulation favoring accumulation in the interstitial space of tumors through enhanced permeation and retention. However, these relatively incompressible nanoparticles are much less echogenic than those with a gas-core. We have worked to characterize and optimize the acoustic response of PLGA-shelled, perfluorooctyl bromide (PFOB) core nanoparticles (radii 70 to 200 nm ; shell-thickness-to-radius ratio 0.25 to 1). Acoustic response (attenuation coefficient, ultrasonic velocity, and relative backscattered intensity) was explored *in vitro* for ranges of concentration, shell-thickness, acoustic pressures and pulse durations (20 to 40 MHz). Modification of the surface chemistry of the polymeric shell with fluorescent, pegylated, or/and biotinylated phospholipids was not associated with apparent response modification. Successful incorporation of paclitaxel in the shell has been achieved but currently the thick-shells of these nanoparticles impede ultrasound-triggered delivery. Although work remains to better adapt liquid-filled nanoparticles for detection and delivery, trials demonstrating nanoparticle detection *in vivo* in mice indicate their potential for accumulation, and detection in tumors.

2:55

**2pBA6. Ultrasound-mediated delivery of bioactive nanobubbles to vascular tissue.** Jonathan T. Sutton, Jason L. Raymond (College of Eng. and Appl. Sci.; Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, 3940 Cardiovascular Ctr., Cincinnati, OH 45267, suttonjt@mail.uc.edu), Michael C. Verleye (College of Eng.; Dept. of Chemical Eng., Notre Dame Univ., Notre Dame, IN), Gail J. Pyne-Geithman (College of Medicine; Dept. of Neurosurgery, Univ. of Cincinnati, Cincinnati, OH), Jack Rubinstein, and Christy K. Holland (College of Medicine; Internal Medicine, Div. of Cardiovascular Diseases, Univ. of Cincinnati, Cincinnati, OH)

Bubble liposomes (BLs) are under development for ultrasound-triggered release of a potent vasodilator within the vasculature. Nano-sized vesicles facilitate this process by enclosing bubbles to enhance ultrasound image contrast, and deliver therapeutic agents. Assessment of drug delivery in living tissue allows for mechanistic pathways to be revealed. In this study, we used a novel *ex vivo* model to assess the vascular effects of ultrasound-mediated delivery of a bioactive gas-nitric oxide (NO)-from nanobubble liposomes. Porcine carotid arteries were excised post-mortem and mounted in physiologic buffer. Vascular tone was assessed in real time by coupling the artery to an isometric force transducer. NO-loaded BLs were infused into the lumen of the artery, which was exposed to 1-MHz pulsed ultrasound, while acoustic cavitation emissions were monitored. Changes in vascular tone were concurrently measured and compared to control and sham NO exposures. Our results demonstrate that ultrasound-triggered NO release from BLs induces potent vasorelaxation within porcine carotid arteries. This approach is a valuable mechanistic tool to assess the bioeffects that NO elicits within the vasculature upon release from BLs exposed to 1-MHz ultrasound.

## *Contributed Papers*

3:15

**2pBA7. Acoustic droplet vaporization is initiated by superharmonic focusing.** Oleksandr Shpak (Phys. of Fluids, Univ. of Twente, Witbreuksweg 379-404, Enschede 7522ZA, Netherlands, o.shpak@utwente.nl), Martin Verweij (Dept. of Imaging Sci. and Technol., Delft Univ. of Technol., Delft, Netherlands), Rik Vos, Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Detlef Lohse, and Michel Versluis (Phys. of Fluids, Univ. of Twente, Enschede, Netherlands)

Acoustically sensitive emulsion droplets composed of a liquid perfluorocarbon have the potential to be a highly efficient system for local drug delivery, embolotherapy or for tumor imaging. The physical mechanisms underlying the acoustic activation of these phase-change emulsions into a bubbly dispersion, termed acoustic droplet vaporization, have not been well understood. The droplets have a very high activation threshold, its frequency dependence does not comply with homogeneous nucleation theory and focusing spots have been observed. We showed that acoustic droplet vaporization is initiated by a combination of two phenomena: highly nonlinear distortion of the acoustic wave before it hits the droplet, and focusing of the distorted wave by the droplet itself. At high excitation pressures, nonlinear distortion causes significant superharmonics with wavelengths below the diameter of the droplet. Because these superharmonics strongly contribute to the focusing effect, the mechanism also explains pressure thresholding effects. In an accompanying paper, mathematical modeling aspects are presented. A proposed model is validated with experimental data captured with an ultra high-speed camera on the positions of the nucleation spots.

Moreover, the presented mechanism explains the hitherto counterintuitive dependence of the nucleation threshold on the ultrasound frequency.

**3:30**

**2pBA8. Subwavelength droplets in nonlinear ultrasound fields: Simulation of focusing effects.** Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., P.O. Box 5046, Delft 2600GA, Netherlands, m.d.verweij@tudelft.nl), Oleksandr Sphak (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands), Hendrik J. Vos, Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Detlef Lohse, and Michel Versluis (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands)

Ultrasound can trigger the evaporation of tiny droplets of emulsified, superheated fluids like perfluorocarbon. This acoustic droplet vaporization (ADV) effect is important because of its potential medical applications. For example, drug-loaded nanodroplets can penetrate the vessel wall and subsequently release their therapeutic load through acoustic activation. Until now, the medical application of ADV has been limited by lack of understanding of the acoustic activation mechanism. In an accompanying paper, a mechanism is proposed that can fully explain the experimentally observed phenomena such as a frequency dependent pressure threshold. The mechanism involves focusing of higher harmonics of the nonlinear activation field. In the current presentation it will be explained how the focusing effect is simulated. A particular problem here is that typical droplet sizes are in the order of micrometers, while typical wavelength sizes are in the order of hundreds of micrometers. This difference in scale will render the traditional analytic solution numerically useless. The problem is avoided by using appropriate expansions of the functions involved. Numerical results are presented that show a focusing effect inside the droplet for a range of incident harmonics. These results have enabled the demonstration of the mechanism in the accompanying paper.

**3:45**

**2pBA9. Characterization of nanometric ultrasound contrast agents with a liquid perfluorocarbon core.** Ksenia Astafyeva (Lab. of Parametric Imaging, Pierre & Marie Curie Univ., 15 rue de l'ecole de medecine, Paris 75013, France, ksenia.astafyeva@gmail.com), Jean-Marc Conoir, Matthieu Guédra (Inst. of Jean Le Rond d'Alembert, Paris, France), Elias Fattal (Institut Galien, Univ. Paris South, Chatenay-Malabri, France), Christine Pepin, Ange Polidori (Univ. of Avignon, Avignon, France), Nicolas Taulier (Lab. of Parametric Imaging, Pierre & Marie Curie Univ., Paris, France), Jean-Louis Thomas (Inst. of NanoSci., Paris, France), Nicolas Tsapis (Institut Galien, Univ. Paris South, Chatenay-Malabri, France), Tony Valier-Brasier (Inst. of Jean Le Rond d'Alembert, Paris, France), and Wladimir Urbach (Lab. of Parametric Imaging, Pierre & Marie Curie Univ., Paris, France)

In this work, we propose a new type of nanometric ultrasound contrast agents (nUCA) with a liquid core and we model their acoustic propagation through their dilute solutions. These capsules have a shell made of a biocompatible polymer or fluorinated surfactants and a liquid perfluorocarbon core to undergo a high lifetime. The capsules are small enough (from 100 nm to 1 $\mu$m) to pass through the tumor endothelium, and they remain stable *in vitro* for several months. Ultrasound attenuation and speed of sound measurements through dilute suspensions of nUCA were carried out from 3 to 90 MHz at various temperatures and concentrations. The acoustic propagation was modeled by combining (i) a dilatational mode taking into account the radial oscillations of the capsules, and (ii) a translational mode of oscillations induced by viscoinertial interaction with the continuous phase. The model makes possible to fit with good accuracy the experiments using values compatible with literature data. Moreover, it reveals information about unknown parameters of the shell: for instance, the viscoelastic shell has to be described as a Maxwell rheological medium. [This work was supported by Emergence-UPMC program.]

**4:00–4:15 Break**

**4:15**

**2pBA10. Ultrasonic propagation in suspensions of encapsulated compressible nanoparticles.** Matthieu Guédra (Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie - Paris 6, 4 Pl. Jussieu, Paris 75252, France, matthieu.guedra@dalembert.upmc.fr), Ksenia Astafyeva (Laboratoire d'Imagerie Paramétrique, Université Pierre et Marie Curie - Paris 6, Paris, France), Jean-Marc Conoir, François Coulouvrat (Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie - Paris 6, Paris, France), Nicolas Taulier (Laboratoire d'Imagerie Paramétrique, Université Pierre et Marie Curie - Paris 6, Paris, France), Jean-Louis Thomas (Institut des Nano-Sci. de Paris, Université Pierre et Marie Curie - Paris 6, Paris, France), Wladimir Urbach (Laboratoire d'Imagerie Paramétrique & Laboratoire de Physique Statistique de l'ENS, Université Pierre et Marie Curie - Paris 6, Paris, France), and Tony Valier-Brasier (Institut Jean le Rond d'Alembert, Université Pierre et Marie Curie - Paris 6, Paris, France)

Dispersion and absorption are examined for dilute suspensions of encapsulated droplets of nanometric size, with typical radii around 100 nm. This new generation of contrast agents is designed for targeted delivery of drugs. Compared to standard contrast agents used for imaging, particles are of smaller size to pass the endothelial barrier, their shell made up of biocompatible material is stiffer to undergo a longer time life and they have a liquid (PFC) instead of a gaseous core. Ultrasound propagation of these suspensions is modeled by combining (i) a dilatational mode of oscillation assuming a compressible shell with a visco-elastic behavior of Kelvin-Voigt or Maxwell type (relaxation), (ii) a translational mode of oscillation induced by visco-inertial interaction with the ambient fluid, and (iii) polydispersion in terms of radius and shell thickness. Influence of the various effects will be examined. Experimental measurements of the dispersion and absorption properties of nanodroplets solutions over the 1–100 MHz frequency range are performed for various temperatures and concentrations. They allow to fit with good accuracy the model properties and estimate some unknown mechanical properties of the shell of the nanodroplets. [Work supported by programme Emergence-UPMC—project NACUNAT and by Inserm/Plan Cancer—project NABUCCO.]

**4:30**

**2pBA11. Dual perfluorocarbon nanodroplets enhance high intensity focused ultrasound heating and extend therapeutic window *in vivo*.** Linsey C. Phillips, Paul S. Sheeran, Connor Puett (Joint Dept. of Biomedical Eng., UNC at Chapel Hill, and NC State Univ., CB 7575, Univ. of North Carolina, Chapel Hill, NC 27599, linsey@email.unc.edu), Kelsie F. Timbie, Richard J. Price (Biomedical Eng., Univ. of Virginia, Charlottesville, VA), G. Wilson Miller (Radiology, Univ. of Virginia, Charlottesville, VA), and Paul A. Dayton (Joint Dept. of Biomedical Eng., UNC at Chapel Hill, and NC State Univ., Chapel Hill, NC)

Perfluorocarbon microbubbles are known to enhance high intensity focused ultrasound (HIFU) ablation by cavitation. However, they can result in superficial skin heating, minimizing their clinical translation. Perfluorocarbon nanodroplets activate only at the higher pressures present at the acoustic focus. We hypothesized that a mixed perfluorocarbon nanodroplet formulation would minimize surface heating while still enhancing ablation. Tissue-mimicking phantoms containing microbubbles or nanodroplets were sonicated (1 MHz, 15 W, 60 s) to assess heating and lesion formation *in vitro*. Microbubbles or nanodroplets were injected into rats ($n = 3$) and HIFU (1 MHz, 15 W, 15 s) was focused into each liver while under MRI guidance. Temperature throughout the liver was tracked by MR thermometry. *In vitro*, microbubbles caused excess surface heating during HIFU, whereas nanodroplets did not. *In vivo*, microbubbles typically circulate for less than 15 min. In comparison, the nanodroplets remained viable in circulation for at least 96 min. HIFU lesions of consistent volume were produced during this time and reached the same maximal temperature rise ($\Delta 550$ °C). In the absence of nanodroplets or microbubbles, significantly greater power (25 W) and twice as much time (30 s) was required to generate an ablation lesion in the liver. These results demonstrate the ability of nanodroplets to more safely shorten ablation procedures.

2p TUE. PM

4:45

**2pBA12. Use of micro- and nano-sized inertial cavitation nuclei to trigger and map drug release from cavitation-sensitive liposomes.** Susan M. Graham, Rachel S. Myers, James Choi, Miriam Bazan-Peregrino (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, ORCRB, Oxford OX3 7DQ, United Kingdom, susan.graham@eng.ox.ac.uk), Leonard Seymour (Dept. of Oncology, Univ. of Oxford, Oxford, United Kingdom), Robert Carlisle, and Constantin C. Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Encapsulation of cytotoxic drugs into liposomes enhances pharmacokinetics and improves passive accumulation in tumors. However, stable liposomes have limited drug release, and thus action, at the target site. This inefficient and unpredictable drug release is compounded by a lack of low-cost, non-invasive methods to map release in real time. We present a new liposomal vehicle that is exclusively triggered by inertial cavitation. Ultrasound exposure of these liposomes in the absence of SonoVue® provided no increase in drug release, whilst with SonoVue® at inertial cavitation pressure levels a substantial (30%) and significant ($p < 0.001$) increase was observed *in vitro*. A 16-fold increase in the level of drug release within tumors was similarly observed in the presence of inertial cavitation following intravenous delivery. Passive acoustic mapping of inertial cavitation sources during delivery was also found to correlate strongly with the presence of release. However, variability in tumor perfusion indicated that uneven distribution of micron-sized SonoVue® may limit this approach. Nano-scale cavitation nuclei, which may more readily co-localize with 140 nm liposomes, were thus developed and showed similar cavitation energies to SonoVue® *in vitro*. These nano-nuclei may ultimately provide a more reliable and uniform way to trigger drug release *in vivo*.

5:00

**2pBA13. Time lapse observation of phase change nano droplet after vaporization stimulated by ultrasound.** Kenji Takehara, Takashi Azuma (BioEng., The Univ. of Tokyo, 7-3-1 Hongou, Bunkyou-ku, Tokyo 113-8656, Japan, k.take531@gmail.com), Kiyoshi Yoshinaka (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Ibaraki, Japan), Satoshi Yamaguchi (Chemistry and BioTechnol., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan), Miyuki Maezawa (Olympus Corp., Shinjuku-ku, Tokyo, Japan), Ichirou Sakuma (Precision Eng., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan), Teruyuki Nagamune (Chemistry and BioTechnol., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan), Shu Takagi, and Yoichirou Matsumoto (Mech. Eng., The Univ. of Tokyo, Bunkyou-ku, Tokyo, Japan)

Small enough to permeate through tumor blood vessel, and can be detect by ultrasound, phase change nano droplet (PCND) have been studied as contrast agents and therapeutic sensitizer for career. To investigate performance for these purpose, we investigated physical behavior of PCND, especially a lifetime of microbubbles generated by ultrasound stimulation. To investigate bubble's behavior after phase change, we observed a time-lapse change of bubbles population with following method. Focused transducer with a frequency of 3.3 MHz was placed in a water bath of 37 degrees. At the focal point, polyacrylamide gel including PCND was placed. Focused hydrophone was placed perpendicularly to the direction of ultrasound propagation. Two kind of ultrasound pulse wave was used; phase change pulse at the beginning and observation pulse at every 500 $\mu$s. The amplitude of scattering signal (SS) reflects the sum of scattering cross-section of bubbles. The time lapse observation of PCND after phase change showed two kinds of behavior, quick and slow decay of scattering signal from the bubbles. The unique increase of SS from the phase-changed bubbles in the monitoring phase was observed. We improved the experimental setup to measure bubble's population properly from various directions using 1-D array transducer and will present the result.

5:15

**2pBA14. Detection of unique acoustic signatures for phase-change contrast agents used in medical imaging and therapy.** Paul S. Sheeran, Karl H. Martin (Joint Dept. of Biomedical Eng., Univ. of North Carolina and North Carolina State Univ., 10 DuxforDC., Durham, NC 27703, pssheeran@gmail.com), Jordan N. Hjelmquist (Dept. of Biomedical Eng., North Carolina State Univ., Raleigh, NC), Terry O. Matsunaga (Dept. of Medical Imaging, Univ. of Arizona, Tucson, AZ), and Paul A. Dayton (Joint Dept. of Biomedical Eng., Univ. of North Carolina and North Carolina State Univ., Chapel Hill, NC)

Phase-change contrast agents (PCCAs) provide a dynamic platform to approach problems in medical ultrasound (US). Upon US-mediated activation, the liquid core vaporizes and expands to produce a gas bubble ideal for US imaging and therapy. In this study, we demonstrate through underlying theory, high-speed microscopy, and US interrogation that PCCAs composed of highly volatile perfluorocarbons (PFCs) exhibit unique acoustic behavior that can be differentiated from tissue and standard microbubble contrast agents. Experimental results show that when activated with short pulses PCCAs will over-expand ($6.3\times$ to $7.7\times$ the droplet diameter, PFC dependent) due to momentum of expansion, and undergo unforced, under-damped radial oscillation while settling to a final bubble diameter ($5.1\times$ to $5.5\times$ the droplet diameter). Oscillation frequency is inversely related to droplet size—near 100 kHz for droplets $\geq 4$ $\mu$m in diameter, and 2.5 MHz for droplets near 500 nm. Results from *in vitro* vessel phantoms using confocal piston transducers with an "activate high" (8 MHz, 2 cycles), "listen low" (1 MHz) scheme show that droplet-specific signals can be detected in both time and frequency domain, and that the magnitude of the acoustic "signature" increases with PFC volatility. These signatures may aid in development of droplet-specific detection techniques.

5:30

**2pBA15. Study of mechanism of sonoporation using lipid bilayer and surface-modified microbubble.** Kodai Hirose, Takashi Azuma (Mech. Eng., The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, khirose@fel.t.u-tokyo.ac.jp), Kiyoshi Yoshinaka (National Inst. of Adv. Industrial Sci. and Technol., Tsukuba, Japan), Akira Sasaki, Shu Takagi, and Yoichiro Matsumoto (Mech. Eng., The Univ. of Tokyo, Tokyo, Japan)

Microbubble enhanced sonoporation is one of gene therapies and expected to be safe, less invasive, and controllability of treatment area, but its induction rate is very low and its mechanism remains to be explained. The objective is to analyze the mechanism and obtain an optimal design. Because microbubble density distribution was unstable during sonication and ultrasound intensity on the cell surface was affected by distribution change through attenuation change during propagation, microbubble distribution should be controlled and localized only near the membrane to realize high reproducibility. An artificial lipid bilayer modified by biotin-avidin to bind microbubbles was used for this purpose. We have three steps: build lipid bilayer, binding with microbubbles, and introduction of sonication and observation system. We built the lipid bilayer with a diameter of 1 mm using Black Lipid Membrane method. With a capacitance measurement with impedance analyzer (NF, ZM2375), the lipid bilayer thickness was confirmed and its duration is more than an hour. Then, we introduced the sonication and observation system to the lipid bilayer, and we are observing a behavior of the lipid bilayer under irradiation of ultrasound. Afterward we will observe that of a lipid bilayer modified of microbubbles.

5:45

**2pBA16. Optimization of acoustic parameters and nanodroplet concentration for spatially controlled, reduced energy high intensity focused ultrasound ablation.** Andrew C. Puett (Biomedical Eng., UNC, 6317 S Bradley Overlook, Wilmington, NC 28403, connorpuett@gmail.com), Linsey C. Phillips, Paul S. Sheeran, and Paul A. Dayton (Biomedical Eng., UNC, Chapel Hill, NC)

Background: Perfluorocarbon (PFC)-nanodroplets (ND) provide cavitation sites when vaporized to microbubbles by acoustic energy and lower the power required to ablate tissue by high intensity focused ultrasound (HIFU). However, control over ablation can be problematic. This study explored vaporization, ablation, and PFC-ND concentration *in vitro* to optimize the acoustic pressure (intensity) and insonation time required for spatially controlled HIFU enhancement. Methods: HIFU (continuous wave; 1MHz; 5–20 s; 2–4 MPa) was applied to albumin-acrylamide gels containing PFC-agent (1:1 mix of volatile decafluorobutane and more stable dodecafluoropentane at $10^5$-$10^7$ ND/mL). Controlled ablation was defined as the production of cigar-shaped lesions centered at the acoustic focus. Results: Vaporization field change from "cigar" to "tadpole" began at $5\times10^5$, $2.5\times10^6$, and $1\times10^7$ ND/mL using 4, 3, and 2 MPa, respectively. The volumes of the microbubble clouds (8–200 $mm^3$) and ablation lesions (1–135 $mm^3$) within them were dependent on acoustic intensity, insonation time, and PFC-ND concentration. Conclusions: Ablation within microbubble clouds of predictable size, shape, and location can be generated in gels containing PFC-ND using intensities $\leq$650 $W/cm^2$. Also, pressures and insonation times can be selected to achieve an ablation lesion of desired size for a given PFC-ND concentration. Demonstrating control is an important step toward developing a useful clinical tool.

6:00

**2pBA17. Design and characterization of biocompatible perfluorocarbon nanodroplets for theragnostic application.** Lucie Somaglino, Ksenia Astafyeva (Laboratoire Imagerie Paramétrique, UMR 7623, CNRS-UPMC, 15, rue de l'école de médecine, EscA 2ème ét., Paris 75005, France, lucie.somaglino@yahoo.fr), Stéphane Desgranges, Ange Polidori, Christiane Contino-Pepin (Institut des Biomolécules Max Mousseron, UMR 5247, CNRS-Université d'Avignon et des Pays de Vaucluse, Avignon, France), Wladimir Urbach, and Nicolas Taulier (Laboratoire Imagerie Paramétrique, UMR 7623, CNRS-UPMC, Paris, France)

We have developed stable emulsions ($\geq$ 3 months) made of nanodroplets (nD) of perfluorocarbon (PFC) dispersed in water to serve as theragnostic agent. nD are stabilized by in house fluorinated surfactants, named FTAC, which chemical structure can be modified to tune their properties. We have characterized nD size distributions (mean diameters from 200 to 600 nm), density, adiabatic compressibility, interfacial tension. US properties of the emulsions have been investigated such as attenuation. Lastly, ultrasonic signals backscattered by nanoemulsions were studied and compared with water to extract signal to noise ratio (SNR), by emitting single negative pulses at $\approx$40 MHz. At similar mean/mode diameters, we showed a strong dependence in SNR values (i) with size distribution, altered by the nature of the surfactant or by a centrifugation/filtration process, (ii) with core nature, and (iii) with nD volume fraction. Besides, hydrophobic drugs such as a thalidomide derivative (with anti-angiogenic properties), have been encapsulated by addition of 10% of triacetin in the nD core. So as to study drug release from nD, a dedicated setup of US cavitation was designed. Cavitation was generated in controlled conditions using a 1 MHz focused transducer (US bursts: 12-22 MPa peak) in a thin wall container immersed in water which temperature and degassed level were kept constant. The generation of cavitation in nD solutions resulted in a strong and reproducible SNR decrease.



TUESDAY AFTERNOON, 3 DECEMBER 2013 MASON, 1:30 P.M. TO 3:10 P.M.

## Session 2pEA

## Engineering Acoustics, Acoustical Oceanography, and Animal Bioacoustics: Bearing Measurement Methods for Small Wideband Sonars

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

**Chair's Introduction—1:30**

### *Invited Papers*

1:35

**2pEA1. Use of Bessel side lobe modulation of frequency modulated pulse bearing estimation.** Kenneth M. Walsh (K + M Engineering Ltd., 51 Bayberry Ln., Middletown, RI 02842, kwalsh4@mindspring.com)

The beam pattern of a circular transducer is a set of frequency dependent Bessel functions. By using a broad band FM pulse and a 1–3 composite transducer, it appears that the modulation in amplitude and phase due to a reception of the high frequency components can be used to estimate the echo's angle off the transducer axis. The 1–3 composite transducer has a simple amplitude and phase structure.

**1:55**

**2pEA2. Modeling of bio-inspired broadband sonar for high-resolution angular imaging.** Jason E. Gaudette (Adv. Acoust. Systems Div., NUWC Div. Newport, 1176 Howell St., B1371/3, Newport, RI, jason.e.gaudette@navy.mil) and James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI)

Echolocating mammals perceive images of targets with hyper-resolution and navigate seamlessly through obstacles in complex acoustic environments. The biological solution to imaging with sound is vastly different from man-made sonar. The most prominent difference is that instead of imaging with narrow beams, bats ensonify a large spatial region and exploit broadband echo information to acoustically focus with about one degree of angular resolution. Angular localization may therefore be redefined as a spectral pattern matching problem. By imaging with wider beams, this remarkable performance requires only a single broadband transmitter and two receive elements. Our computational modeling work has led to new insight into the salient spatial information encoded by the bat's auditory system. Although theoretically not required, spatial localization performance increases with the aid of highly complex baffle structures such as those found in biological sonar. Replicating these bio-inspired baffle structures and acoustic processing techniques in man-made systems can reduce sonar array aperture requirements by a factor of 100 or more for a variety of both aerial and underwater acoustic sensing applications. Recent modeling results are presented along with progress toward the design of a compact bio-inspired sonar system for high-resolution imaging. [Work supported by ONR and NUWC Newport.]

**2:15**

**2pEA3. Front-looking and side-looking receiving beams for biosonar imaging and flight guidance.** James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI 02912, james_simmons@brown.edu)

The biosonar broadcasts of big brown bats are very broadly beamed over $\pm 60°$-$120°$, thus ensonifying virtually all of the objects in the surrounding sonar scene. However, masking-release results indicate that off-side clutter is rejected outside of a significantly narrower beam of $\pm 15°$. Recent experiments establish that clutter interference with target imaging is prevented by sensing the lower amplitude of FM2 relative to FM1 in clutter echoes and forming poorly-focused images of the clutter. Removal of clutter from perception does not address the need to guide flight in complex surroundings bounded by clutter such as vegetation. A separate mechanism used for guidance may be located in the midbrain, where exclusively contralaterally sensitive neurons respond to target range and direction, the ingredients for following the flow of surrounding objects as they slide past the flying bat. A review of neurophysiological results from different laboratories suggests that the big brown bat's biosonar system may contain different receiver pathways—a forward-looking system for target imaging and classification, and a side-looking system for following the surrounding clutter. [Work supported by ONR and NSF.]

**2:35**

**2pEA4. Improving direction-sensing by multibeam sonar.** Gerard Llort-Pujol (Image and Information Processing Dept., Institut Mines Télécom - Télécom Bretagne, Brest, France), Kenneth G. Foote (Woods Hole Oceanographic Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu), and Christophe Sintes (Image and Information Processing Dept., Institut Mines Télécom - Télécom Bretagne, Brest, France)

The complexity of multibeam sonar systems makes their beamforming susceptible to amplitude and phase distortion, e.g., due to environmental changes. A form of calibration is usually performed *in situ* over a flat area to ensure flatness in the resulting sonar image. However, lack of detailed knowledge of individual channel performance prevents application of an amplitude-weighting function such as the Blackman or Hamming type, which could otherwise be used to reduce the impact of sidelobes without damaging signal quality. Two radically different solutions are proposed: application of the Vernier principle [G. Llort-Pujol, Oceans 2006 MTS/IEEE Conf. Proc. (Quebec City, Canada, 2008); C. Sintes *et al*., Oceans 2011 MTS/IEEE Conf. Proc. (Waikoloa, HI, 2011)] to interferometry performed at low grazing angles, and performance of a standard-target calibration [K. G. Foote *et al*., J. Acoust. Soc. Am. **117**, 2013 (2005)] to measure the two-way sensitivity of individual channels directly. The two methods, which are also applicable to sidescan sonar, are elaborated.

## *Contributed Paper*

**2:55**

**2pEA5. Bearing measurements using a compact wideband sensor by forming three dipoles.** David A. Brown (BTech Acoustics LLC, ATMC, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

This paper summaries the development of a compact sensor for determining the bearing angle of an incoming signal using trinary acoustic dipoles. The approach is a kin to the using orthogonal dipoles an omni-reference, which is common in DIFAR sonobuoy. Symmetrically processing three dipoles can offer advantages in sensitivity and extended bandwidth. Analytical and experimental results on the TRI-phase Bearing Estimator, or TRIBE, are presented.

## Session 2pED

## Education in Acoustics: Take 5's

Jack Dostal, Chair
*Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109*

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

TUESDAY AFTERNOON, 3 DECEMBER 2013 CONTINENTAL 1, 1:00 P.M. TO 3:50 P.M.



## Session 2pMU

## Musical Acoustics and Signal Processing in Acoustics: Digital Musical Instruments

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

### *Invited Papers*

**1:00**

**2pMU1. Interaction design and the active experience of music.** David Wessel (Music CNMAT, Univ. of California Berkeley, 1750 Arch St., Berkeley, CA 94709, davidwessel@me.com)

Music search engines, play list generators, streaming audio, and portable players have taken much of the focus of music technology. The emphasis is on delivery and experiencing music is by playback, playback while jogging or while working about the house, and sadly even while studying. In this talk in the hope of providing an antidote I will examine the role of bodily action in the experience of music and the importance of human computer interaction design in the development of computationally based musical instruments. Central are gestural interfaces and their mapping to musical material. Special emphasis will be given to designing for expression, for exploration and discovery, and to a musical practice that involves a coordinated balance of software development and daily bodily engagement with one's instrument.

**1:20**

**2pMU2. LinnStrument and other new expressive musical controllers.** Roger Linn (Roger Linn Design, 1147 Keith Ave., Berkeley, CA 94708, rl@rogerlinndesign.com)

Roger Linn will demonstrate his LinnStrument, a controller for musical performance that captures three dimensions of movement for each touch, polyphonically, in order to provide more expressive control of software music synthesis. In addition, he will show videos of other similar new instruments and compare the unique approaches taken by each designer.

**1:40**

**2pMU3. Sound synthesis for a brain stethoscope.** Chris Chafe, Juan-Pablo Caceres, and Michael Iorga (Stanford Univ., CCRMA/Music, Stanford, CA 94305, cc@ccrma.stanford.edu)

Exploratory ascultation of brain signals has been prototyped in a project involving neurologists, real-time EEG and techniques for computer-based sound synthesis. In a manner similar to using a stethoscope, the listener can manipulate the location being listened to. Sounds which are heard are sonifications of electrode signals. We present a method for exploring sounds from arrays of sensors as sounds which are useful for distinguishing brain states. The approach maps brain wave signals to modulations characteristic of human voice. Computer-synthesized voices "sing" the dynamics of wakefulness, sleep, seizures, and other states. The goal of the project is to create a recognizable inventory of such vocal "performances" and allow the user to probe source locations in the sensor array in real time.

**2:00**

**2pMU4. Grafting acoustic instruments and signal processing: Creative control and augmented expressivity.** Dan Overholt (Media Technol., Aalborg Univ. Copenhagen, A.C. Meyers Vaenge 15, Copenhagen SV. 2450, Denmark, dano@create.aau.dk)

In this study, work is presented on a hybrid acoustic/electric violin. The instrument has embedded processing that provides real-time simulation of acoustic body models using DSP techniques able to gradually transform a given body model into another, including extrapolations beyond the models to explore interesting new timbres. Models can include everything from various violin bodies to guitars, sitars with their sympathetic strings, and even physically impossible acoustic bodies. The development also presents several practical approaches to sensor augmentation and gestural playing techniques that can be applied to bowed-string and other acoustic instruments, in order to provide immediate creative control over the possibilities offered by DSP. The study has focused on augmenting the expressivity of the violin toward finding novel timbral possibilities, rather than a goal of simulating prior acoustic violins with high fidelity. The opportunity to control a virtually malleable body while playing, i.e., a model that changes reverberant resonances in response to player input, results in interesting audio effects. Other common audio effects can also be employed and simultaneously controlled via the musician's movements. For example, gestural tilting of the instrument is tracked via an embedded Inertial Measurement Unit (IMU), which can be assigned to alter parameters such as the wet/dry mix of an octave-doubler or other effect, further augmenting the expressivity of the player.

**2:20**

**2pMU5. Saxophone fingering identification.** Tamara Smyth (Music, Univ. of California San Diego, 9500 Gilman Dr., MC 0099, La Jolla, CA 92093-0099, trsmyth@ucsd.edu) and Marjan Rouhipour (Computing Sci., Simon Fraser Univ., Surrey, BC, Canada)

The focus of this work is to identify the tonehole configuration or "fingering'' applied by a player during performance, using only the signal recorded at the bell. Because of a player can use alternate fingerings/overblowing to produce a given frequency, detecting the sounding pitch only reduces the possible candidates—it does not produce a unique result. Several recordings of a professional saxophonist playing notes using all fingerings are considered, and several higher level features are explored for distinguishing between a fundamental and an overblown note. In the latter case, it is observed that during the attack portion of the note, the spectral centroid is usually lower, there is greater inharmonicity and increased pitch instability. Combining these heuristics with the detection of subharmonics has yielded excellent results in detecting overblown notes. With the possible fingerings being greatly reduced by this preprocessing, more computationally expensive statistical methods may be employed for a more accurate estimation of the actual fingering applied. To this end, the recorded sound is calibrated to that produced by a reed model coupled to a waveguide that is informed by an acoustic measurement of the player's saxophone configured with each usable fingering.

**2:40**

**2pMU6. The Faust Synthesis Toolkit: A set of linear and nonlinear physical models for the Faust programming language.** Romain Michon (Dept. of Music, Ctr. for Comput. Res. in Music and Acoust., 660 Lomita Court, Stanford, CA 94305-8180, rmichon@ccrma.stanford.edu)

The Faust Synthesis ToolKit is a set of virtual musical instruments written in the Faust programming language and based on waveguide algorithms and on modal synthesis. Most of them were inspired by instruments implemented in the Synthesis ToolKit (STK) and the program SynthBuilder. Our attention has partly been focused on the pedagogical aspect of the implemented objects. Indeed, we tried to make the Faust code of each object as optimized and as expressive as possible. Some of the instruments in the Faust-STK use nonlinear allpass filters to create interesting and new behaviors. Also, a few of them were modified in order to use gesture data to control the performance. A demonstration of this kind of use is done in the Pure Data program. Finally, the results of some performance tests of the generated C++ code are presented.

**3:00**

**2pMU7. Drawn to sound: An audio visual musical instrument using custom electronics and magnetometer.** John Granzow and Hongchan Choi (CCRMA, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, johknee5@gmail.com)

Drawings are amplified through a resonant surface and transmitted via microphone to custom software hosted on an embedded linux computer. An HMC-5883L magnetometer is used to modulate the signal according to the position of the pencil (equipped with magnetic sleeve). 3D vectors are derived from the magnetometer using a custom Arduino library. We project this vector into a 2D plane to get magnitude or distance between the sensor and the magnetized pencil as well as the heading angle. This implementation gives us 2D polar coordinates such that the position of the pencil can be used to vary the audio output. The transformation of the raw drawing sound is excited with proximity to the magnetometer due to the sensors exponentially increasing sensitivity to the magnetic field. Resulting drawings often contain both visually motivated marks as well as gestures that are made for sound such as dark scribbled regions where a desired timbre or pitch shift is repeated throughout a performance. This presentation will discuss hardware design, the implementation of our custom software and circuitry, how these components combine for a compelling performance platform as well as areas where we seek improvement.

## *Contributed Papers*

**3:20**

**2pMU8. The Stingray embedded acoustic instrument.** Edgar J. Berdahl (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

Many traditional acoustic musical instruments are convenient to use: a performer picks one up, provides an energetic excitation, and the resulting sound radiates immediately. No wires, protocols, or software updates are ever required. The same cannot be said for the vast majority of prior digital musical instruments. The present work addresses how to endow a digital musical instrument with the convenience, look, feel, and personality of a traditional acoustic musical instrument via enclosure prototyping techniques, audio amplification, and embedded computation. For example, the Stingray is an embedded acoustic instrument. Although its battery needs to be occasionally charged, it otherwise can give the impression of a traditional acoustic musical instrument. The control inputs for the Stingray include a piano keyboard and force-feedback motorized faders. The faders allow the performer to interact expressively with the sound, while the piano keyboard enables the precise selection of notes. If desired, the Stingray can be programmed using physical models, including models of hypothetical acoustic instruments that would be infeasible to build physically. In this configuration, the Stingray expressively transforms the performer's gestures into radiated sound in an energy-conserving manner.

**3:35**

**2pMU9. Towards a bendable circuit model of the Casio SK-1 keyboard.** Kurt J. Werner (CCRMA, Stanford Univ., 223 Ayrshire Farm Ln., Apt. 205, Stanford, CA 94305, kwerner@ccrma.stanford.edu)

The Casio SK-1 keyboard was introduced in 1985 and synthesizes sampled and built-in sounds via pulse-code modulation and additive ("harmonic") synthesis. Initially important as one of the first home keyboards with sampling capabilities, the SK-1 has become one of the most popular instruments for circuit bending, the process of creatively modifying or augmenting sound-producing electronic devices. I create a parameterized component-level software model of the analog circuitry of the Casio SK-1, with applications to archiving and preserving its historic sound, expanding its basic behavior through circuit-bent modifications and extensions, and providing circuit benders with a resource for informed bending. Throughout, special attention is paid to creating models in terms of the circuit's component values. The SK-1's Percussion, Bass, and Chord Filters are modeled and analyzed in continuous-time as transfer functions and discretized via the bilinear transform. The non-linear (including diodes and transistors) Envelope/Pitch Mixing Circuit and Melody Filter are analyzed with linearizing simplifications to elucidate design intent, and modeled as ordinary differential equations to capture their behavior accurately. A review of techniques for the numerical solution of ordinary differential equations follows. A model of the internal speaker's impulse response and estimated static non-linearity rounds out the project.

## Session 2pNS

## Noise, Physical Acoustics, and Structural Acoustics and Vibration: Launch Vehicle Acoustics

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

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

R. Jeremy Kenny, Cochair
*NASA, M.S. ER42, Bldg. 4203, Marshall Space Flight Ctr., Huntsville, AL 35812*

### *Invited Papers*

**1:00**

**2pNS1. Far-field acoustic modeling of rocket noise to determine community impacts.** Michael M. James, Alexandria R. Salton, and Micah Downing (Blue Ridge Research and Consulting, LLC, 15 W. Walnut St., Ste. C, ASHEVILLE, NC 28801, michael.james@blueridgeresearch.com)

The emerging commercial space market is generating interest in commercial launch site ("spaceport") development around the United States. FAA regulations require all new spaceports to acquire a launch site operator's license, which is considered a Federal action subject to environmental review. Potential noise impacts are evaluated based on FAA Order 1050.1E, Change 1, Environmental Impacts: Policies and procedures, which include the assessment of DNL and may be supplemented with additional acoustical metrics. These supplemental metrics may range from speech interference to structural damage impacts. Extensive studies and research have examined the appropriateness of these metrics in relation to aircraft operations. However, the differences between these acoustic sources and operational modes stress the need for computer models and impact criteria specific to launch vehicles. Further measurements and research are needed to improve rocket source characterization, long-range sound propagation of high amplitude waveforms through complex atmosphere, and environmental and community impacts. The evolving nature of the regulatory environment surrounding rocket noise warrants a renewed focus on appropriate noise modeling and impact criteria to determine potential conflicts with launch noise.

**1:20**

**2pNS2. Use of a large microphone array to identify noise sources during a rocket engine test firing and a rocket launch.** Jayanta Panda (Experimental Aero-Phys. Branch, NASA Ames Res. Ctr., M.S. 260-1, Moffet Field, CA 94035, jayanta.panda-1@nasa.gov), Robert N. Mosher, and Barry J. Porter (Experimental Aero-Phys. Branch, Aerosp. Computing, Inc., Mountain View, CA)

A 70 microphone, 10 ft×10 ft, microphone phased array was built for use in the harsh environment of rocket launches. The array was setup at NASA Wallops launch pad 0A during a static test firing of Orbital Sciences' Antares engines, and again during the first launch of Antares vehicle. It was placed 400 ft away from the pad, and was hoisted on a scissor lift 40 ft above ground. The data sets provided unprecedented insights into rocket noise sources. The duct exit was found to be the primary source during the static test firing; the large amount of water injected beneath the nozzle exit quenched all other sources. The noise maps during launch were found to be time-dependent. As the engines came to full power and became louder, the primary source switched from the duct inlet to the duct exit. Further elevation of the vehicle caused spilling of the hot plume, resulting in a distributed noise map covering most of the pad. As the entire plume emerged from the duct, and the on-deck water system came to full power, the plume itself became the loudest noise source. These noise maps will help to improve the sound suppression system for future launches.

**1:40**

**2pNS3. Acoustic measurements of the Epsilon rocket at liftoff.** Tatsuya Ishii, Seiji Tsutsumi, Kyoichi Ui, Hideshi Oinuma (JAXA, 7-44-1 Jindaiji-higashi-machi, Chofu, Tokyo 182-8522, Japan, ishii.tatsuya@jaxa.jp), Yutaka Ishii (Bruel & Kjaer Japan, Tokyo, Japan), and Kei Wada (Science Service, Inc., Tokyo, Japan)

Launch vehicles generate intense acoustic field caused by the enormous thrust power during liftoff, and this acoustic field leads to harmful payload vibration. Japan Aerospace Exploration Agency (JAXA) plans to launch a new solid propellant rocket, Epsilon, in 2013. This three-stage rocket utilizes a reliable first stage motor of the JAXA's H-2 rocket, SRB-A. Since the SRB-A is expected to cause excessive acoustic load, a countermeasure was required to mitigate the acoustic feedback to the vehicle. Computational works proposed a launch pad structure to attenuate the Mach wave radiation from the plume, and the acoustic wave generated by the plume impinging to the flame deflector. In the previous ASA conference, the authors introduced scale model tests using 1:42 scale rocket motors and the launch pad models. The scale model tests clarified the acoustic benefit of the launch pad structure. The tested geometry of the launch pad structure was adopted to the full-scale structure. Acoustic measurements are planned in the first launch of the Epsilon rocket

in order to evaluate the acoustic influence with this newly constructed launch pad structure. The measurement setup and a brief review of the measurements (if possible) are discussed.

**2:00**

**2pNS4. Scale model thruster acoustic measurement results.** Magda B. Vargas and Robert J. Kenny (MSFC, NASA, M.S. ER42, Bldg. 4203, Marshall Space Flight Ctr., Huntsville, AL 35812, magda.b.vargas@nasa.gov)

The Space Launch System (SLS) Scale Model Acoustic Test (SMAT) is a 5% scale representation of the SLS vehicle, mobile launcher, tower, and launch pad trench. The SLS launch propulsion system will be comprised of the Rocket Assisted Take-Off (RATO) motors representing the solid boosters and four Gas Hydrogen (GH2) thrusters representing the core engines. The GH2 thrusters were tested in a horizontal configuration in order to characterize their performance. In phase 1, a single thruster was fired to determine the engine performance parameters necessary for scaling a single engine. A cluster configuration, consisting of the four thrusters, was tested in phase 2 to integrate the system and determine their combined performance. Acoustic and overpressure data was collected during both test phases in order to characterize the system's acoustic performance. The results from the single thruster and 4-thuster system are discussed and compared.

**2:20**

**2pNS5. Near-field/far-field study of the end-effects regime produced by large area ratio nozzles.** Raymundo M. Rojo, Charles E. Tinney, Woutijn J. Baars (Aerosp., Univ. of Texas at Austin, 701 28th St., Apt. 407, Austin, TX 78712, raymundo.rojo46@gmail.com), and Joseph H. Ruf (NASA Marshall Space Flight Cente, Huntsville, AL)

Vibro-acoustic loads emanating from large area ratio rocket nozzles during start-up can be catastrophic to the launch system and payload. This study quantifies a particular feature referred to as the "end-effects regime", which is considered the largest source of vibro-acoustic loading during start-up [Nave and Coffey, AIAA Paper 1973-1284]. In this experiment, data acquired during the start-up sequence of several full-scale rocket engines are compared to the laboratory-scale measurements of a thrust-optimized parabolic-contour nozzle conducted in a fully anechoic chamber. The laboratory studies encompass both static and dynamic wall pressures measured inside the nozzle, as well as far-field acoustic surveys. The event produced during the "end-effects regime" was successfully reproduced in the sub-scale model, and was characterized in terms of its mean, variance, and skewness, as well as the spectral properties of the signal obtained by way of time-frequency analyses. The intensity and characteristic frequency of the event of interest are discussed through a comparison of the nominal values for the full-scale and sub-scale system and whether they obey with standard scaling laws.



**2:40**

**2pNS6. Evaluation of Japanese current primary launch vehicle liftoff acoustic environment change due to launch pad facility modifications.** Hiroki Ashida, Makoto Hirai (Aerosp. Systems, Mitsubishi Heavy Industries, Ltd., 10, Oye-cho, Minato-ku, Nagoya City, Aichi 455-8515, Japan, hiroki1_ashida@mhi.co.jp), Keita Terashima, and Takumi Ujino (Space Transportation Mission Directorate, Japan Aerosp. Exploration Agency, Ibaraki, Japan)

H-IIA, Japanese primary launch vehicle, has been successfully launched 21 flights with a success rate of 95.4%. During 12 years of its operational phase, extensive acoustic measurements on the vehicle and on the ground have been conducted to refine conventional prediction methods and to evaluate the effect of vehicle/pad configuration changes. In this presentation, the evaluation of the effects of major pad configuration modification on H-IIA liftoff acoustic environment is presented. The effect of water injection is also evaluated.

**3:00–3:15 Break**

**3:15**

**2pNS7. Numerical study on acoustic generation of a supersonic jet impinging to deflectors.** Seiji Tsutsumi, Ryoji Takaki (JEDI, JAXA, 3-1-1 Yoshinodai, Chuuou, Sagamihara, Kanagawa 252-5210, Japan, tsutsumi.seiji@jaxa.jp), Koji Okamoto (Dept. of Adv. Energy, Univ. of Tokyo, Kashiwa, Japan), and Susumu Teramoto (Dept. of Aeronautics and Astronautics, Univ. of Tokyo, Bunkyo-ku, Japan)

Acoustic wave generated from a $M=1.8$ ideally expanded jet impinging on flame deflectors is investigated numerically in order to obtain the knowledge of flame deflector design to mitigate the acoustic loading on launch systems and payloads. Mechanism of acoustic wave generated from a 45-degree-inclined flat plate placed 5D downstream from the nozzle exit is analyzed first, and it is revealed that source of the acoustic wave is located where shock waves is formed due to the jet impingement. In this study, effect of the deflector shape will be discussed for better understanding the acoustic generation mechanism.

**3:35**

**2pNS8. Towards jet acoustic prediction within the Launch Ascent and Vehicle Aerodynamics framework.** Jeffrey A. Housman (Appl. Modeling and Simulation Branch, NASA Ames Res. Ctr., M.S. N-258, Moffett Field, CA 94035, jeffrey.a.housman@nasa.gov), Christoph Brehm (Appl. Modeling and Simulation Branch, Sci. and Technol. Corp., Moffett Field, CA), and Cetin Kiris (Appl. Modeling and Simulation Branch, NASA Ames Res. Ctr., Moffett Field, CA)

Understanding the acoustic environment generated during lift-off is critical for successfully designing new space vehicles. In order for modeling and simulation tools to effectively assist in the development of the vehicles, validation must be performed on simplified model problems. In this paper, time-accurate implicit large eddy and detached eddy simulations coupled with a linear acoustic propagation method are applied to a Mach 1.8 perfectly expanded jet impinging on a flat plate at 45 degrees. The Launch Ascent and Vehicle Aerodynamics (LAVA) code used to simulate this problem is a high-fidelity unsteady simulation tool for modeling fluid dynamics, conjugate heat transfer, and acoustics. A detailed description of the linear acoustic propagation tool is provided. The narrow band far-field

sound pressure levels predicted using LAVA are compared to existing experimental data. POD and spectral methods are applied to analyze the noise sources due to coherent flow structures and jet impingement. Grid and time-step sensitivity studies are performed to assess the spatial and temporal requirements for accurate jet acoustic simulation. Sensitivity of the predicted far-field sound pressure levels to position of the acoustic propagation surface is also assessed.

**3:55**

**2pNS9. Intensity-based approach to characterize near-field acoustic environments of space flight vehicles.** Michael M. James, Alexandria R. Salton (Blue Ridge Research and Consulting, LLC, 15 W. Walnut St., Ste. C, Asheville, NC 28801, michael.james@blueridgeresearch.com), Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

An intensity-based measurement probe has been developed to measure the magnitude, directivity, and spectral content of near-field rocket source noise. An array of the intensity-based measurement probes was deployed in a static test firing of the GEM-60 at Alliant Techsystems in Promontory, UT, in 2012. The probes were positioned along the shear layer of the rocket motor plume to enable a comparison of the resultant source characterization obtained via sound power type measurement approaches to traditional acoustic measurement methodologies. The measurement results demonstrate that the intensity-based probes advance the measurement and characterization of the near-field acoustic environment of rockets. Moreover, intensity-based acoustic data provides an important role in formulating more realistic sound source models, improving acoustic load estimations, and aiding in the development of the next-generation space flight vehicles via improved measurements of noise near the rocket plume.

**4:15**

**2pNS10. Methods for estimating acoustic intensity in rocket noise fields.** Derek C. Thomas, Benjamin Y. Christensen, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., N223 ESC, Provo, UT 84602, derekcthomas@gmail.com)

The acoustic field produced by launch vehicles is difficult to measure and characterize. Acoustic intensity measurements provide more information per measurement location than pressure measurements and are therefore interesting for the characterization of rocket noise fields. The extreme environment associated with a rocket requires a robust intensity probe while the large size of the source and the high-amplitude, highly nonlinear behavior of the system produce a signal with significant low and high frequency components. Thus, the probe must also provide accurate results over a large frequency band. The bandwidth limitations of the standard method for estimating acoustic intensity, the p-p finite difference method, motivated the development of an alternative method for intensity estimation. The new phase and amplitude gradient estimation (PAGE) method will be presented and compared to the standard p-p method. Specific features of the rocket noise field that can be leveraged to improve the bandwidth of the intensity estimates will also be discussed.

**4:35**

**2pNS11. Acoustic intensity estimates from a solid rocket motor test firing.** Benjamin Christensen, Derek C. Thomas, and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., 539E 300N, Provo, UT 84606, ukeben@gmail.com)

Acoustic measurements of a static GEM-60 solid motor test firing were taken as part of a continuing effort to characterize the aeroacoustic source regions and noise environment around launch vehicles. Multiple 2D intensity probes, consisting of four coplanar microphones, were used in the measurement. Two intensity estimation techniques have been applied to the data: the finite difference p-p method, and the new phase and amplitude gradient estimation (PAGE) method. We will present and compare results from both methods and compare to measurements made at past test firings. It appears that the PAGE method for estimating acoustic intensity provides usable results over a larger frequency bandwidth.

## Session 2pPA

# Physical Acoustics and Structural Acoustics and Vibration: Acoustics of Pile Driving: Measurements, Models, and Mitigation II

Kevin M. Lee, Cochair
*Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758*

Karl-Heinz Elmer, Cochair
*OffNoise-Solutions GmbH, Leinstr. 36, Neustadt a. Rbge. 31535, Germany*

### *Invited Papers*



**1:30**

**2pPA1. Effects of pile driving on fishes.** Arthur N. Popper (Univ. of Maryland, Biology/Psych. Bldg, College Park, MD 20742, apopper@umd.edu), Michelle B. Halvorsen (Battelle-Pacific Northwest National Lab., Sequim, WA), Thomas J. Carlson (ProBioSound, Holmes Beach, FL), Michael E. Smith (Univ. of Western Kentucky, Bowling Green, KY), and Brandon M. Casper (Naval Submarine Medical Res. Lab., Groton, CT)

We examined the physiological effects of impact pile driving on fishes using a specially designed tube that allows replication of the far field acoustic conditions of impulsive stimuli. Studies show that the received signal levels needed to result in onset of effects is a combination of single strike sound exposure level (SELss) and cumulative sound exposure level (SELcum), although not in an equal energy relationship. In contrast to current interim regulations, which indicate that the onset of physiological effects occurs at 187 dB SELcum, our experimental results for six species of fishes showed that the onset of physiological effects, none of which produced mortality, was at about 207 dB SELcum. This onset SELcum had to be at least 7–10 dB higher to result in effects that could potentially be mortal. Additional studies showed that fishes can recover from the effects of pile driving and that a fish species without a swim bladder showed no effects, at least up to an SELcum of 216 dB. Investigations on the effects of pile driving on sensory hair cells of the inner ear, an analog for hearing loss, showed that damage only occurred at SELcum that are substantially higher than onset of other effects.

**1:50**

**2pPA2. A Monte Carlo approach to determining marine mammal exposure risk to long term marine piling operations.** Paul A. Lepper (School of Electron., Elec. and Systems Eng., Loughborough Univ., Loughborough LE113TU, United Kingdom, p.a.lepper@lboro.ac.uk), Stephen P. Robinson, Pete D. Theobald, and Tanja Pangerc (Acoust. Group, National Physical Lab., Teddington, United Kingdom)

The expansion of offshore renewable developments, primarily offshore wind, has led to widespread use of large scale percussive piling for foundation construction. In UK waters alone, over 900 foundations installed up to 2013, mostly mono-piles with extensive upscaling of developments planned for the next decade. Pile diameters range from a few meters to greater than 6 m and penetration depths of 20–30 m. These piles are typically percussively driven with several thousand hammer strikes over periods of several hours with individual strike hammer energies in excess of 1900 kJ occasionally used and reported per strike underwater Sound Exposure Level source levels of 215 dB re 1 $\mu Pa^2 s\text{-}m^2$. Potential exists for injury to occur from cumulative sound exposure to repetitive but lower level signals at greater range. If simple receptor behaviors are assumed (static, fleeing, etc.) exposure over time to an entire pile construction sequence can be estimated. These models have been extended using a Monte Carlo approach to model long term, entire wind farm construction scenarios with repetitive foundation construction periods of 24–36 h. The statistical distribution of exposure risk is modeled as well as analysis of the sensitivity of behavioral responses to potential impact effects such as habitat exclusion.

**2:10**

**2pPA3. Noise mitigation systems (NMS) for reducing pile driving noise: Experiences with the "big bubble curtain" relating to noise reduction.** Michael A. Bellmann and Patrick Remmers (itap GmbH, Marie-Curie-Str. 8, Oldenburg 26160, Germany, bellmann@itap.de)

For the offshore wind farm Borkum West II in the German North Sea the Noise Mitigation System (NMS) "Big Bubble Curtain" was used during pile driving activities. Within this project systematically variations of different influencing factors on noise reductions such as air volume, nozzle hose sizes, distance of nozzle hoses, etc., were investigated. Additionally the "Big Bubble Curtain" is currently in use for different other OWF in the German North Sea. Therefore, the "Big Bubble Curtain—BBC" is at the moment one of the most investigated NMS under offshore condition. Within this presentation, experiences and results of the above listed projects will be shown and discussed.

**2:30**

**2pPA4. Evaluation of hydro sound and vibration measurements during the use of the Hydro-Sound-Damper (HSD) at the wind farm "London Array".** Benedikt Bruns (Technische Universität Braunschweig, Beethovenstrasse 51 b, Braunschweig 38106, Germany, b.bruns@tu-bs.de)

Since some years a noise prevention concept for the protection of marine animals exists in Germany. Based on that, the acoustic underwater noise from the pile driving at offshore wind farms is required to be less than 160 dB (SEL) at a distance of 750 m. This value, however, is often exceeded so that the use of a soundproofing system is necessary. The Hydro-Sound-Damper (HSD) is a new, versatile method to reduce the noise during offshore pile driving. To achieve this, elements of different sizes and materials are used, which are fixed to fishing nets. The principle of operation and the effectiveness of these HSD elements were investigated in the laboratory and in situ under offshore conditions at the world's largest offshore wind farm "London Array." During the offshore tests thorough measurements were performed which metered the propagation of the hydro sound and the vibrations of the sea floor at various distances and directions. The evaluation of these data led to very promising results concerning underwater noise reduction. This article describes the theory and implementation of the HSD at "London Array" and focuses on the interpretation of the data from the hydro sound and vibration measurements.

## *Contributed Papers*

**2:50**

**2pPA5. Efficient application of encapsulated bubbles and foam elements to mitigate offshore piling noise.** Karl-Heinz Elmer (OffNoise-Solutions GmbH, Leinstr. 36, Neustadt a. Rbge. 31535, Germany, karl-heinz.elmer@t-online.de)

The very high noise levels of offshore piling noise during the installation of offshore wind converters are dangerous to marine life like harbor porpoises and fishes. Encapsulated bubbles and foam elements are successfully used to reduce the very high noise levels. There are different physical effects such as impedance mismatch, resonance and scattering effects and material damping effects that are responsible to the measured noise reduction between about 10 dB (SEL) and 23 dB (SEL). The radiation of underwater noise from the pile surface, the propagation and the reflections of the radiated waves are investigated together with the influence of encapsulated bubbles and foam elements in the near field of the pile. Only a very small part of the impact energy is radiated directly from the pile into the surrounding water. Most of the hammer energy is driven into the ground. A small part from this is radiated indirectly from the ground into the water. The efficiency of encapsulated bubbles and foam elements in underwater noise mitigation is investigated by theoretical studies, laboratory measurements, and by offshore measurements with different elements and combinations.

**3:05**

**2pPA6. Dependence of resonance frequencies and attenuation for large encapsulated bubbles on bubble wall thickness and bubble fill-material.** Gregory Enenstein, Preston S. Wilson, and Kevin M. Lee (Mech. Eng. Dept. and Appl. Res. Lab., The Univ. of Texas at Austin, 4700 W Guadalupe St #A-437, Austin, TX 78751, gregenenstein@gmail.com)

Arrays of large encapsulated bubbles are currently under development for the purpose abating low-frequency anthropogenic underwater noise from various sources including marine pile driving and oil and gas exploration and production. An existing predictive model [Church, J. Acoust. Soc. Am. **97**, 1510–1521 (1995)], which was originally intended to describe propagation through suspensions of microbubbles used as ultrasound contrast agents, was previously found to be in good agreement with resonance frequency and attenuation measurements using large encapsulated with radii on the order of 10 cm [Lee *et al*., J. Acoust. Soc. Am **132**, 2039 (2012); Lee and Wilson, Proceedings of Meetings on Acoustics **19**, 075048 (2013)]. For the current study, both laboratory and lake experiments were performed on large encapsulated bubbles to investigate the dependence of the bubbles' resonance frequencies and attenuation on the bubble wall thickness. Additionally, laboratory measurements were made to investigate the effects on encapsulated bubble resonance frequencies and damping using bubble fill-materials other than air, and a lake experiment was then performed to relate these effects on the damping to the attenuation provided by arrays of such bubbles. [Work supported by AdBm Technologies.]

**3:20–3:35 Break**

**3:35**

**2pPA7. Acoustic measurements and modeling of air-filled, underwater resonator cavities.** Laura Tseng, Kevin M. Lee (Appl. Res. Laboratories: The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, Ltseng@utexas.edu), Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Mark S. Wochner (AdBm Technologies, Austin, TX)

This paper investigates the near-resonance acoustical properties of submerged air-filled resonators intended for use in an underwater noise abatement system. These resonators are a potential alternative to encapsulated bubbles. The resonators are similar to Helmholtz resonators in shape and design, but without a neck, consisting of underwater inverted air-filled cavities with rigid walls. A finite element model was developed to investigate the acoustic behavior of the resonators near their resonance frequencies, and based on the results of the model, physical realizations of the resonators were designed and fabricated for testing. Experiments were performed with the resonators in a closed water-filled tank operated in the long wavelength limit [J. Acoust. Soc. Am. **132**, 2039 (2012)], where their resonance frequencies and *Q*-factors were measured using the technique described by Leighton *et al*., [J. Acoust. Soc. Am. **112**, 1366–1376 (2002)]. Comparison between the results from the measurements and modeling will be discussed. [Work supported by Shell Global Solutions.]

**3:50**

**2pPA8. Radiated sound from a scale-model pile submerged in a two--layer medium.** Kevin M. Lee, Todd A. Hay, Taylor W. Weaver, Preston S. Wilson, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu)

Underwater noise due to both marine pile driving and offshore wind farm operation is not only radiated directly from the pile into the water, but also from the seabed surrounding the pile. While there is much interest in mitigating the noise from these activities, a better understanding of the source mechanisms and propagation is needed to determine optimal strategies for noise abatement. A recent analytical model of the acoustic field radiated by submerged piles includes radiation from the pile directly into the water and into a stratified viscoelastic sediment as well as propagation into a shallow water waveguide from both the direct and sediment radiation paths [Hay *et al*., Proceedings of Meetings on Acoustics **19,** 070038 (2013)]. As a step towards validating this model, scale-model experiments were conducted in the high kilohertz frequency range with a model pile consisting of a mechanically excited metallic tube inserted into a laboratory tank filled with two stratified layers to simulate the water/sediment interface. Measurements of the acoustic field in the experiment are compared with the model predictions, and the relevance of these results to implementing noise abatement strategies will be discussed. [Work supported by ARL:UT IR&D.]

4:05

**2pPA9. On the resonance frequency of an ideal arbitrarily shaped bubble.** Kyle S. Spratt, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 4307 Caswell Ave., Apt. E, Austin, TX 78751, sprattkyle@gmail.com), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Mark S. Wochner (AdBm Technologies, Austin, TX)

Large encapsulated bubbles have recently been described for use in abating low-frequency anthropogenic underwater noise [J. Acoust. Soc. Am. **130**, 3325–3332 (2011)], and the use of encapsulation allows for the possibility of bubbles that are nonspherical in their equilibrium state. For the purpose of more accurately determining such bubbles' resonance frequencies, a lumped-element model of the linear oscillation of an ideal, arbitrarily shaped gas bubble in an incompressible liquid is presented. The corresponding boundary-value problem required to predict the resonance frequency of the bubble is seen to be equivalent to a classic problem in electrostatics [J. Acoust. Soc. Am. **25**, 536–537 (1953)]. Predictions made for the resonance frequency of prolate and oblate spheroidal bubbles using this model are tested against a finite-element model of the full acoustic scattering problem. Particular attention is then paid to the case of an ideal toroidal bubble of arbitrary thickness, and predictions made for the resonance frequency of such a bubble using the lumped-element approach are compared to a finite-element model of the full acoustic scattering problem as well as to existing approximate models for the dynamics of thin toroidal bubbles. [Work supported by AdBm Technologies, LLC and the ARL:UT McKinney Fellowship in Acoustics.]

4:20

**2pPA10. The effect of sediment stiffness on the piling pulse—Results from a wave-equation analysis model.** Michael A. Wood and Victor F. Humphrey (Fluid Dynam.. and Acoust. Group, Inst. of Sound and Vib. Res., Faculty of Eng. and the Environment, Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, maw1v07@soton.ac.uk)

The rapid expansion of wind farms in UK waters has led to increased concern of the anthropogenic noise emitted into the sea due to piling. The piling process generates high levels of noise capable of propagating over long distances in the water column; it is important that this noise can be predicted, and the likely environmental impact determined. The model presented in this work comprises a method known as Wave Equation Analysis of Piles. This technique models the stress-wave as it propagates along the pile using a finite-difference approach. Additionally, the plastic nature of the sediment is modeled at both the pile wall and pile toe. Previous work has shown that by considering the radial expansion of the pile, the results of this model may be coupled to an acoustic model. The effect of the sediment on the pile motion has previously received little attention. The results show that although the first downward-going pulse is independent of the sediment parameters, reflected pulses are affected. The calculations show that there exists an impedance condition based on the soil parameters: for low soil stiffnesses the pulse is inverted at the toe end, whereas for higher stiffnesses no inversion is seen.

4:35

**2pPA11. Physical models and improvement of bubble curtain for the suppression of underwater noise from a pile drive.** Alexander Sutin and Hady Salloum (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Man-made sounds in ocean and inland water environments become biologically significant when they affect the ability of animals and fish to survive and reproduce. Extremely strong sounds produced by pile driving can highly exceed the environmental safety limits and application of bubble curtain is one of the most effective methods for sound suppression. We present several physical models explaining sound suppression by bubble curtain and discuss the methods of improving bubble curtain efficiency. The physical models of sound suppression by a bubble curtain include: (a) Estimation of bubble curtain impedance leading to decreasing of the pile drive acoustic coupling with surrounding bubbly water and (b) theoretical model for the estimation of sound attenuation by resonance bubbles. The developed models were analyzed for the optimization of the pile drive sound suppression. Several methods for generating a bubble curtain with small bubbles and bubbles with varied sizes are considered. We also suggest a way for improving the efficiency of bubble curtains by increasing the lifetime of the bubbles using bubble coating. Coated micro bubbles are widely used as ultrasound contrast agents in cardiology. One of the simplest ways for micro bubble coating is the passing of bubbles through oil.



TUESDAY AFTERNOON, 3 DECEMBER 2013 CONTINENTAL 2/3, 1:00 P.M. TO 5:00 P.M.

## Session 2pPP

## Psychological and Physiological Acoustics: Building a Stairway to Hearing

Sridhar Kalluri, Chair

*Starkey Hearing Res. Ctr., Starkey Hearing Technologies, 2150 Shattuck Ave., Ste. 408, Berkeley, CA 94704*

### *Contributed Papers*

1:00

**2pPP1. Testing and extending the Woodworth model.** William M. Hartmann (Phys. & Astronomy, Michigan State Univ., East Lansing, MI 48824, hartman2@msu.edu) and Neil L. Aaronson (Natural Sci.& Mathematics, Richard Stockton College of NJ, High Bridge, NJ)

The Woodworth model and formula for interaural time difference is frequently used as a standard in physiological and psychoacoustical studies of binaural hearing for humans and other animals. It is a frequency-independent, ray-tracing spherical head model that is expected to agree with an exact diffraction model in the high-frequency limit. The predictions by the Woodworth model for antipodal ears and for incident plane waves are compared with the predictions of the exact model as a function of frequency to quantify the discrepancy when the frequency is not high. In a second calculation, the Woodworth model is extended to arbitrary ear angles, both for plane-wave incidence and for finite point-source distance. This extended Woodworth model leads to different formulas in six different regions defined by ear angle and source distance. It is noted that the characteristic cusp in Woodworth's well-known function comes from ignoring the longer of the two paths around the head in circumstances when the longer path is actually important. This error can be readily corrected.

1:15

**2pPP2. Minimum audible angle at the subjective front during listener's active and passive head rotation.** Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Katahira 2-1-1, Aoba-ku, Sendai 981-0942, Japan, yoh@riec.tohoku.ac.jp), Akio Honda (Tohoku Fukushi Univ., Sendai, Japan), Kagesho Ohba, Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Yukio Iwaya (Tohoku Gakuin Univ., Sendai, Japan)

Listener's head movement, particularly horizontal rotation, effectively improves sound localization acuity (Wallach, 1939; Thurlow, 1967; Kawaura, 1989). However, few findings have been obtained concerning sound localization during head rotation. In the present study, we directly investigated the minimum audible angle (MAA) at the front during horizontal rotation. A sound stimulus (30-ms noise burst) was presented from a loudspeaker of a circular array (r = 1.1 m), with a loudspeaker separation of 2.5 degrees. The listener, sitting at the center of the circle, was asked to answer whether the sound stimulus was presented from the left or right of the subjective front (2AFC). We designed three listening conditions, static, active rotation and passive rotation. In the static condition, listeners were asked to keep their heads still. For the active rotation condition, listeners were asked to rotate their heads. Meanwhile, for the passive rotation condition, listeners sitting on a revolving chair were rotated by an experimenter. In the latter two conditions, the test stimulus was triggered during head movement. Results showed the MAA to deteriorate significantly in the two rotation conditions. This implies that the improvements in sound localization due to head motion could be explained by the multiple-look model (Viemeister, 1991).

1:30

**2pPP3. Difference of the perceived auditory space between walking and passive self-motion.** Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Hideaki Terashima (Graduate School of Information Sci., Tohoku Univ., Sendai, Japan), Wataru Teramoto (Dept. of Comput. Sci. and Systems Eng., Muroran Inst. of Technol., Muroran, Japan), Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Jiro Gyoba (Graduate School of Arts and Letters, Tohoku Univ., Sendai, Japan)

We are investigating how auditory space was represented during linear self-motion (Teramoto *et al*., 2013). Several studies have suggested that whether the listener's motion is active or passive affected sound localization (Hirahara *et al*., 2013). In the present study, therefore, we set up three conditions: active motion condition, passive motion condition, and no motion condition. In active motion condition, observers were walking straight ahead. In passive motion condition, observers were transported forward by a robotic wheelchair. During the self-motion, a short noise burst was presented from one of the loudspeakers which were aligned parallel to the traveling direction when the listener's coronal plane reached the location of one of the speakers (null point). The listeners indicated the direction in which the sound was perceived relative to their coronal plane (i.e., a two-alternative forced-choice task). The results of experiment showed that the sound position aligned with the subjective coronal plane was displaced compared with the null point. However, there was no significant difference between auditory space in active and passive motion conditions. This result suggests only action of the kinetic system during self-motion and planning and execution of voluntary movement would not affect perceived auditory space.

1:45

**2pPP4. Sound source localization from tactile aids for unilateral cochlear implant users.** Xuan Zhong, Shuai Wang, Michael Dorman, and William Yost (Dept. of Speech and Hearing Sci., Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85287, xuan.zhong@asu.edu)

The present research asks whether two tactile aids with directional microphones, by providing additional inter-channel level information and etc., could help unilateral cochlear implant (CI) localize sound sources. For normal hearing subjects, sound source localization based on tactile vibration cues alone can be as accurate as auditory localization in the frontal horizontal plane (Gescheider, 1970). CI users may as well benefit from additional tactile aids just as normal hearing people do. The current study uses two bone-anchored hearing aids (BAHA) as sources of tactile vibration. The two BAHAs, bonded together by a special gadget to maintain a particular distance and angle, both have directional microphones, and are programed so that one point to the front-left side and the other to the front-right side. Unilateral CI users voluntarily participated in the experimental study. Wide band noise stimuli were presented at 65 dB SPL. The subjects hold one BAHA in each hand and do localization tasks with (1) CI only and (2) with CI and tactile sensation combined. Preliminary data shows CI users can get some benefit from the additional information provided by tactile aids in 360 degree localization (45 degree spacing) on the horizontal plane.

2:00

**2pPP5. Characterization of the available feedback gain margin at two device microphone locations, in the fossa triangularis and Behind the Ear, for the light-based contact hearing device.** Suzanne C. Levy, Daniel J. Freed (EarLens Corp., 200 Chesapeak Dr., Redwood City, CA 94063, slevy@earlenscorp.com), and Sunil Puria (EarLens Corp., Stanford, CA)

Assistive devices compensate for hearing impairment by amplifying sounds with gain, which is limited by acoustic feedback. The light-based Contact Hearing Device (CHD) provides amplification to 10 kHz by mechanically vibrating the umbo with a wireless Tympanic-Contact Actuator (TCA). Driving the eardrum mechanically generates a pressure wave, which travels laterally down the ear canal and produces feedback. Placing the microphone in the fossa triangularis (FT) may preserve more natural acoustic cues than the BTE location, although it may reduce available feedback gain margin (FGM). Thirteen subjects with bilateral mild-to-severe hearing impairment were fit with CHDs (26 ears). The TCA was driven with light-pulses and the feedback pressure was measured at the FT and above the pinna at the BTE microphone locations. The mean FGM varied from 32 to 48 dB and 38 to 60 dB for the FT and BTE microphone locations, respectively. FGM was lowest in the 3–5 kHz range and highest at about 7 kHz (below 1 kHz FGM is not measurable). The STD of FGM varied from 5.3 to 13 dB due individual anatomies. A microphone at the BTE has 6 to 12 dB additional FGM over the FT location, and allows a broader inclusion range to fit patients with amplification to 10 kHz.

2:15

**2pPP6. A long-overdue review of empirical uncertainties in the "fatigue" found through simultaneous dichotic loudness balance.** Lance Nizami (Independent Res. Scholar, Wilkie Way, Palo Alto, CA 94306, nizamii2@att.net)

SDLB uses equalization of the loudnesses of stimuli at the two ears (one stimulus intermittent, one constant) to measure the alleged "fatigue" over time of the loudness of the constant stimulus (Hood, 1950). Hood found 50 dB of "fatigue", and SDLB remained influential for over a quarter-century. However, the interpretation of SDLB was long questioned; recently, a novel model uniting the physiology and the behavior emerged (Nizami 2012, Int. Soc. Psychophys., Ottawa, Canada), and others independently re-measured "fatigue". Classically, the stimulus waveforms at the two ears were similar, sometimes identical, permitting two equalization techniques when the stimuli coincided in time: centering of the sound between the ears (lateralization), or matching the loudness contributions from each ear (loudness-matching). Further, "fatigue" was habitually expressed as across-listeners averages. But careful scrutiny reveals that (1) over the "fatiguing" duration, lateralization may give way to loudness-matching, (2) dedicated loudness-matching may nonetheless yield only half as much "fatigue" as lateralization, and (3) the standard deviation of "fatigue" can be half its mean value, such that some listeners would not have "fatigued." In sum, the magnitude of "fatigue" is remarkably uncertain, and is likely to remain so until auditory physiology is compellingly integrated into explanations of SDLB.

**2:30**

**2pPP7. Influence of measurement method and context of presentation on the loudness difference between increasing and decreasing intensity sounds.** Emmanuel Ponsot, Patrick Susini (IRCAM, 1 Pl. Igor Stravinsky, Paris, France, patrick.susini@ircam.fr), and Sabine Meunier (LMA, CNRS, UPR 7051, Aix-Marseille Univ, Centrale Marseille, Marseille, France)

Four experiments were conducted to assess the loudness of both increasing and decreasing intensity sounds using different methods and context of presentation as between-subjects factors. In Exp 1 and Exp 2, loudness was assessed directly by using magnitude estimation procedures, with increasing and decreasing sounds presented respectively either in the same block or in separate blocks. In the other two experiments, loudness was measured by the mean of pairwise comparisons. While increasing and decreasing sounds were compared with each other in Exp 3, they were compared respectively with constant-intensity sounds in Exp 4. Two-intervals, 2AFC interleaved-adaptive procedures were used to prevent from potential biases. As a result, very similar trends were observed in the four experiments. In particular, the loudness difference between increasing and decreasing sounds always felt within the same range: decreasing intensity sounds need to be about 3 dB louder that increasing sounds to be perceived with equal loudness. This study thus indicates that this loudness asymmetry actually corresponds to a true perceptual effect and is not due to any experimental bias, since a clear consistency across the results was found using different measurement methods and context of presentation.

**2:45–3:00 Break**

**3:00**

**2pPP8. Effect of sound duration on loudness estimates of increasing and decreasing intensity sounds.** Emmanuel Ponsot, Anne-Laure Verneil, and Patrick Susini (IRCAM, 1 Pl. Igor Stravinsky, Paris 75004, France, ponsot@ircam.fr)

The influence of sound duration on global loudness of non-stationary stimuli was investigated. Loudness of 2 and 6-s increasing and decreasing intensity sounds with different ranges of intensity-variation was assessed using a magnitude estimation procedure. Results once again uphold the existence of a loudness difference between the two patterns: while they only differ in their temporal profile, increasing sounds were perceived louder than decreasing sounds. In addition, global loudness estimates were increased with duration for the two types of sounds, and a small but significant interaction occurred between type and duration. A contrast analysis revealed that while global loudness of increasing and decreasing sounds raised with duration in a similar way in the case of low and moderate intensities (below 75 dB SPL), global loudness was significantly more affected by duration with increasing than with decreasing intensity profiles for high-intensity stimuli. This result suggests the existence of an underlying memory process combined with a "peak-end rule" as being responsible for the loudness asymmetry typically observed between the two types of intensity-pattern being judged [Susini *et al.* (2010). "End level bias on direct loudness ratings of increasing sounds" J. Acoust. Soc. Am. EL **128**(4), 163–168].

**3:15**

**2pPP9. A novel bispectral approach to study cochlear non-linearities in transient evoked otoacoustic emissions.** Gabriella Tognola, Alessia Paglialonga, Emma Chiaramello, and Stefano Moriconi (Inst. of Electronics Computer, and Telecommun. Eng. ISIB, CNR It. Natl. Res. Council, Piazza L. da Vinci, 32, Milan 20133, Italy, gabriella.tognola@polimi.it)

A new approach, based on bispectral analysis, is proposed to study non-linearity in cochlear active mechanisms, as evaluated in transient evoked otoacoustic emissions (TEOAEs). Based on an ad hoc formulation of the bispectral analysis, specifically developed in this study to fit the particular features of TEOAEs, this innovative method detects non-linearity by extracting quadratic frequency couplings (QFCs) in TEOAE recordings. The method directly estimates non-linear TEOAE components, which may not be detected by conventional spectral analysis and can thus provide important information useful for better modeling cochlear active mechanisms and to detect sub clinical cochlear dysfunctions. The method was characterized with synthesized TEOAEs as a function of the main TEOAE parameters and then used to analyze TEOAEs recorded in normal hearing adults and full-term neonates to: (i) obtain normative data; (ii) to evaluate test retest repeatability of non-linear interaction mechanisms; and (iii) to investigate the influence of stimulus intensity on non-linear interaction mechanisms. Results revealed that most of the energy of non-linear components is located in the 1–4 kHz range and that the test retest repeatability of non-linear interaction mechanisms is high. The factor that most affects non-linearities is stimulus intensity.

**3:30**

**2pPP10. Finite element model of feed-forward/feed-backward amplification in the mouse cochlea.** Joris Soons (Biomedical Phys., Univ. of Antwerp, 496 Lomita Mall, Stanford, California 94305, jsoons@stanford.edu), Sunil Puria, and Charles R. Steele (Mech. Eng., Stanford Univ., Stanford, CA)

Thousands of hair cells in the organ of Corti, situated along the basilar membrane (BM), detect displacements due to sound input. For low input sounds, these displacements are amplified by active outer hair cells (OHCs). A proposed theory is the feed-forward/feed-backward mechanism for the OHC amplification where an expanding hair cell gives a forward push through the Deiters Cells and a backward pull on the BM through the Phalangeal process. Previously this was implemented mathematically using WKB theory (Yoon *et al.* 2011, Biophys. J.). In the present work, we explicitly modeled this as a Y-shaped arrangement of the OHC-DeitersCell-PhallangealProcess to form a building block using beam elements in a finite element formulation. These Y-shaped blocks were chained together to construct a single-row organ-of-Corti model from the base to apex, coupled to the BM and scalae fluid, of a mouse cochlea. The OHC force is proportional to the shear on the BM. For a 10 kHz stapes input tone, the passive BM reaches a peak gain of about 26 dB. For the active case the BM gain increases to 58 dB and shifts apically by about 0.6 mm. These results are consistent with physiological measurements in several other living animals.

**3:45**

**2pPP11. Should the acceptable noise level be considered to be an acceptable noise range?** Jonas Brännström, Lucas Holm, Tobias Kastberg (LogopeDC, phoniatrics, and Audiol., Lund Univ., Clinical Sci. Lund, Lund SE-22185, Sweden, jonas.brannstrom@med.lu.se), and Steen J. Olsen (Dept. of Otorhinolaryngology, Head and Neck Surgery, Univ. Hospital, Rigshospitalet, Copenhagen, Denmark)

The acceptable noise level (ANL) test is used to quantify the amount of competing background noise (BNL) that a listener is willing to accept when listening to speech at the most comfortable level (MCL). ANL is calculated by subtracting the BNL from the MCL. Most studies show large intersubject ANL variability and a few also demonstrate large intrasubject variability. Very few predictor variables for ANL have been identified and it has been proposed that the ANL depends on an inherent characteristic of the listener. However, some of the variability seems to depend on poor precision of the ANL test. After removing the effect of poor precision, some variability still remains. One possible explanation for these findings may be that the ANL is not a single level but a range of levels. Using recent data, this presentation examines the notion of an acceptable noise range.

**4:00**

**2pPP12. Effects of speakers' language background on speech perception in adults.** Mark A. Dame, Harisadhan Patra, Petula C. Vaz (Audiol. & Speech Pathol., Bloomsburg Univ. of PA, 226 CEH, 400 E 2nd St., Bloomsburg, PA 17815, hpatra@bloomu.edu), and Biswajit Ray (Electronics Eng. Technol., Bloomsburg Univ. of PA, Bloomsburg, PA)

With an increasingly changing sociolinguistic environment in the U.S., clinicians are challenged to accurately evaluate individuals' speech perception under realistic everyday listening conditions. This study investigated how contextual linguistic information, speakers' language background, speech-rate, and background noise individually and interactively affect listeners' ability to recognize sentences. Ten normal-hearing American English native speakers, aged 20–22 years were recruited as listeners. Eight normal-hearing native speakers of four different languages (American English, Chinese-Mandarin, German, and Spanish) were recruited as speakers. These speakers read 120 sentences, 60 high-predictability and 60 low-predictability sentences from the revised Speech in Noise test (Bradlow and Alexander, 2007), which served as test stimuli. The listeners reported the target word of each sentence presented either in quiet or in multitalker babble. Results revealed that noise, high speech-rate, and lack of contextual cues could have significant adverse affects on listeners' test scores. Speakers' language background also had significant effects on listeners' performance. Specifically, listeners had the most difficulty in perceiving the Chinese-Mandarin speakers, followed by the Spanish, and German speakers; more so in multitalker babble than in quiet. Further studies are warranted since results may have implications for audiology diagnosis and rehabilitation to address effective everyday communication.

**4:15**

**2pPP13. Training Mandarin-speaking amusics to recognize pitch direction: Pathway to treat musical disorders in congenital amusia?** Fang Liu (Dept. of Linguist and Modern Lang., The Chinese Univ. of Hong Kong, Rm. G36, Leung Kau Kui Bldg., Shatin, N.T., Hong Kong, China, fangliufangliu@gmail.com), Cunmei Jiang (Music College, Shanghai Normal Univ., Shanghai, China), Tom Francart (ExpORL, Dept. of NeuroSci., KU Leuven, Leuven, Belgium), Alice H. Chan (Div. of Linguist and Multilingual Studies, School of Humanities and Social Sci., Nanyang Technolog. Univ., Singapore, Singapore), and Patrick C. Wong (Dept. of Linguist and Modern Lang., The Chinese Univ. of Hong Kong, Hong Kong, China)

Congenital amusia is a lifelong disorder of musical perception and production that has been hypothesized as due to impaired pitch direction recognition. This study investigated whether amusics could be trained to identify pitch direction in speech and music, and if so, whether enhanced pitch direction recognition would benefit musical processing in amusia. Eighteen Mandarin-speaking amusics and 18 matched controls were evaluated using the Montreal Battery of Evaluation of Amusia (Peretz *et al*., 2003) and tested on two psychophysical pitch threshold tasks for identification of pitch direction in the Mandarin syllable /ma/ and its piano tone analog. Subsequently, nine of the eighteen amusics undertook a 10-session training program on pitch direction identification in /ma/ and piano tone. Compared with those untrained, trained amusics demonstrated significantly improved thresholds for pitch direction identification in speech syllables [pretest: 2.92 (4.63) semitones; posttest: 0.16 (0.06)], and marginally significant improvement in pitch direction identification thresholds for piano tones [pretest: 5.20 (4.56); posttest: 0.89 (1.46)]. These findings suggest that pitch sensitivity of individuals with congenital amusia could be improved through auditory training, providing evidence for neural plasticity in the amusic brain, which may lead the way to other rehabilitative programs for treating this musical disorder.

**4:30**

**2pPP14. Neuromagnetic beta-band oscillation for rhythmic processing induced by subjectively accented structure.** Takako Fujioka (CCRMA (Ctr. for Comput. Res. in Music and Acoust.), Dept. of Music, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305-8180, takako@ccrma.stanford.edu), Laurel J. Trainor (Dept. of Psych., Neurosci. and Behavior, McMaster Univ., Hamilton, ON, Canada), and Bernhard Ross (Rotman Res. Inst., Baycrest, Toronto, ON, Canada)

Musical rhythm facilitates synchronized body movements and schema-based, predictive timing perception. Our previous magnetoencephalography (MEG) study demonstrated that beta-band (~20 Hz) activity in bilateral auditory cortices shows synchronized modulation that predicts the time point of the next beat (Fujioka *et al*. 2009, 2012). Furthermore, after finger tapping to a different musical meter such as a march or waltz (every 2nd or 3rd beat), the broadband evoked response from auditory cortex differentiates the metric conditions (Fujioka *et al*., 2010). Here we examined how beta-band activity indexed subjective metrical perception during listening to unaccented beats (1) after listening to acoustically accented beats, and (2) after finger tapping with either the left or right index finger. The auditory cortices showed beat-synchronized modulation in line with the previous studies in both march and waltz conditions. However, distinction between down-beat and up-beat positions was stronger in march than in waltz condition, with symmetrical activities in the left and right auditory cortices. This contrast between the two metric conditions was stronger in the side contralateral to the tapping finger in the tapping condition. This suggests that contribution of auditory cortices to metric processing depends on both its timing structure and rhythmic movement in the contralateral hemisphere.

**4:45**

**2pPP15. On the learnability of auditory concepts.** Ronaldo Vigo, Mikayla Barcus, Yu Zhang, and Charles Doan (Ohio Univ., 200 Porter Hall, Athens, OH 45701, vigo@ohio.edu)

The field of categorization and concept learning research has been dominated by findings involving visual stimuli. Among these findings is the learning difficulty ordering of the family of category structures associated with visual categorical stimuli consisting of four objects defined over three dimensions. This ordering has been observed numerous times in several rigorous studies (Kruschke, 1992; Love *et al*., 2004; Nosofsky *et al*., 1994; Shepard *et al*., 1961; Vigo, 2011a, 2013a. 2013b) and has been influential in shaping current theories of conceptual behavior. In recent work, we have freed the field from this visual bias by examining the learnability of auditory categorical stimuli that are instances of the aforementioned structures. We found that, in general, for auditory categorical stimuli the learning difficulty ordering of these structures is somewhat different from that of their visual counterparts. However, we also found that this difference may be explained and accurately predicted by a simple encoding mechanism proposed in generalized invariance structure theory (GIST; Vigo, 2013b). We view this result as evidence in support of the proposition that a single basic conceptual system underlies the acquisition of both auditory and visual concepts.

## Session 2pSA

## Structural Acoustics and Vibration: Computational Structural Acoustics

Jerry W. Rouse, Cochair
*Analytical Structural Dynam., Sandia National Labs., P.O. Box 5800, Albuquerque, NM 87185*

Timothy F. Walsh, Cochair
*Computational Solid Mechanics and Structural Dynam., Sandia National Labs., PO Box 5800, M.S. 0380, Albuquerque, NM 87185*

**Chair's Introduction—1:00**

### *Invited Papers*



**1:05**

**2pSA1. Material identification in frequency-domain coupled acoustic-structure interaction using an error in constitutive equation functional.** Wilkins Aquino, James Warner, and Manuel Diaz (Civil and Environ. Eng., Duke Univ., Hudson Hall, Durham, NC 27708, wa20@duke.edu)

This work focuses on the inverse identification of linear elastic material parameters in the context of frequency-domain, coupled acoustic-structure interaction. The approach postulates the inverse problem as an optimization problem where the solution is obtained by minimizing a modified error in constitutive equation (MECE) functional. The MECE functional measures the discrepancy in the constitutive equations that connect kinematically admissible strains and dynamically admissible stresses while incorporating the measurement data as an additional quadratic penalty term. The method is formulated generally for the recovery of material properties in a coupled acoustic-structure system using solid displacement and/or fluid pressure measurement data. Numerical results demonstrate that the proposed methodology can identify the spatial distribution of elastic moduli from partial and noisy measurements taken in either the fluid or solid subdomains.

**1:25**

**2pSA2. A parallel Helmholtz solver for acoustics and structural acoustics.** Clark R. Dohrmann and Timothy F. Walsh (Computational Solid Mech. & Structural Dynam., Sandia National Labs., P.O. Box 5800, M.S. 0380, Albuquerque, NM 87185-0380, crdohrm@sandia.gov)

In this talk, we present a parallel Helmholtz solver for the solution of acoustic and structural acoustic problems in the frequency domain. The solver is based on overlapping Schwarz domain decomposition concepts for parallelism, and artificial structural damping is introduced in the preconditioner to deal with resonant or near resonant conditions at the subdomain level. An efficient method to solve problems over a range of frequencies is also described. Numerical examples are presented for acoustic and structural acoustic problems, as well as a study of solver performance for higher order finite element discretizations.

**1:45**

**2pSA3. Computational methods for the interior structural acoustics of small spaces.** Karl Grosh, Yizeng Li (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu), and Robert Littrell (Baker-Calling, Inc., Santa Monica, CA)

In some biomechanical systems and micro-electro-mechanical systems (MEMS), the interaction of a viscous compressible fluid confined in a space bounded in part by a flexible structure is of central importance. Two specific examples are MEMS microphones (condenser or piezoelectric) and the cochlea. In both the manmade and biological acoustical sensor, the interior space is typically smaller than an acoustic wavelength, and a successful design involves trade-offs between sensitivity, bandwidth, and noise (including thermal, mechanical, electrical, or channel generated noise); the latter two criteria depend critically on the viscous and thermal forces in the system. A direct numeric approach to modeling viscous and thermal effects is often prohibitively expensive, as boundary layers must be resolved in the mesh. In this talk, we will present approximate methods that enable the inclusion of viscothermal effects in a computational framework. In particular, a variational approach amenable to inclusion in a finite element based code will be presented. Results for this method, which retains the accuracy of a Navier-Stokes formulation with the computational cost of a standard scalar acoustic formulation, will be given along with the limitations of the method and a discussion of alternative numerical approaches.

## *Contributed Papers*

**2:05**

**2pSA4. Convolution formulations for non-negative intensity in a plane.** Earl G. Williams (Acoust. Div., Naval Res. Lab., Code 7106, 4555 Overlook Ave., Washington, DC, DC 20375, earl.williams@nrl.navy.mil)

New spatial convolution formulas for a variant of the active normal intensity in planar coordinates have been derived that use measured pressure or normal velocity near-field holograms to construct a positive-only (outward) intensity distribution in the plane, quantifying the areas of the vibrating structure that produce radiation to the far-field. This is an extension of the outgoing-only (unipolar) intensity technique recently developed for arbitrary geometries by Steffen Marburg. The method is applied independently to pressure and velocity data measured in a plane close to the surface of a point-driven, unbaffled rectangular plate in the laboratory. It is demonstrated that the sound producing regions of the structure are clearly revealed using the derived formulas and that the spatial resolution is limited to a half-wavelength. A second set of formulas called the hybrid-intensity formulae are also derived which yield a bipolar intensity using a different spatial convolution operator, again using either the measured pressure or velocity. Using the experiment results it is shown that the velocity formula yields the classical active intensity and the pressure formula an interesting hybrid intensity that may be useful for source localization. Computations are fast and carried out in real space without Fourier transforms into wavenumber space. [Work supported by the Office of Naval Research.]

**2:20**

**2pSA5. Minimization of the sound radiated by a curved underwater panel.** Micah R. Shepherd and Stephen A. Hambric (Appl. Res. Lab, Penn State Univ., PO Box 30, M.S. 3220B, State College, PA 16801, mrs30@psu.edu)

The sound radiated by a curved underwater panel excited by a point drive is minimized using an evolutionary strategy. The panel thickness is varied in incremental strips along the length of the panel to obtain the optimal configuration. The panel weight and sound radiation are minimized simultaneously using a single weighted objective function. The weighting coefficient is then varied to obtain a Pareto front describing the competing nature of the two objectives. The optimal designs are compared for each weighting coefficient illustrating the tradeoff between minimizing weight and radiated sound power. Normal modes and radiation efficiency curved are also compared. When the optimizer favors minimizing weight, the element thickness is uniformly minimized. Conversely, when the optimizer favors minimizing sound power, the optimal design accepts energy less easily and becomes an inefficient radiator.

**2:35**

**2pSA6. Vibroacoustic modeling of ventilation window system with internal partial partitions.** Xiang Yu and Li Cheng (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon 00000, Hong Kong, li.cheng@polyu.edu.hk)

Domestic buildings exposed to traffic and environmental noise has always been a major concern. Among various noise control measures, ventilation windows offer appealing features by simultaneously allowing good sound attenuation and air ventilation. However, the modeling of such systems is challenging in that the simulation tools should cope with the complexity of the system; to reach reasonably high frequency range; and to offer the flexibility needed for system optimization. In the present study, the sound transmission through a double-glazed ventilation window is investigated using the patch transfer function method along with a handy treatment of the air aperture and micro-perforated elements. Numerical results show the sound insulation performance can be improved with internal partial partitions, which can be either solid or micro-perforated.

**2:50–3:05 Break**

**3:05**

**2pSA7. Sound field alteration through cavity shape design using a Wavelet-Galerkin Method.** Su Zhang (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong) and Li Cheng (Dept. of Mech. Eng., The Hong Kong Polytechnic Univ., Hung Hom, Kowloon 00000, Hong Kong, li.cheng@polyu.edu.hk)

The general problem of internal sound field prediction along with the manipulation of the cavity boundary shape to alter the sound pressure distribution is dealt with in this paper. Owing to the compactly supported orthogonal property of the wavelet and its extraordinary fitting capability, Daubechies wavelet scaling function is used as a global basis function to expand the unknown sound pressure under the general Galerkin framework. The proposed formulation is shown to offer high accuracy on the numerical calculation of the sound pressure field with the use of a remarkably small number of messing points. A genetic based shape optimization algorithm is proposed and demonstrated. As an example, an enclosure with an inner rigid acoustic screen is investigated. By optimizing the shape of the screen, the sound pressure level within a chosen area is successfully reduced. Results show the remarkable potentials of the proposed approach as a topology optimal tool for the general inner sound field problems.

**3:20**

**2pSA8. A comparison of perfectly matched layers and infinite elements for exterior Helmholtz problems.** Gregory Bunting (Computational Solid Mech. and Structural Dynam., Sandia National Labs., 709 Palomas Dr. NE, Albuquerque, NM 87108, gbunting@purdue.edu), Arun Prakash (School of Civil Eng., Purdue Univ., West Lafayette, IN), and Timothy Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., West Lafayette, Indiana)

Perfectly matched layers and infinite elements are commonly used for finite element simulations of acoustic waves on unbounded domains. Both involve a volumetric discretization around the periphery of an acoustic mesh, which itself surrounds a structure or domain of interest. Infinite elements have been a popular choice for these problems since the 1970s. Perfectly matched layers are a more recent technology that is gaining popularity due to ease of implementation and effectiveness as an absorbing boundary condition. In this study, we present massively parallel implementations of these two techniques, and compare their performance on a set of representative structural-acoustic problems on exterior domains. We compare the ability of these methods to absorb acoustic waves on ellipsoidal domains with waves of relatively high oblique angles of incidence. We also examine the conditioning of the linear systems generated by the two techniques by examining the number of Krylov-iterations needed for convergence to a fixed solver tolerance. [Sandia National Laboratories is a multi-program laboratory managed and operated by Sandia Corporation, a wholly owned subsidiary of Lockheed martin Corporation, for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-AC04-94AL850000.]

**3:35**

**2pSA9. A plane wave method for modeling acoustic variables in cavities.** Matthew Kamrath (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, kamrath64@gmail.com) and Gary Koopmann (KCF Technologies, Inc., State College, PA)

The pressure field within a cavity that contains a homogeneous fluid is modeled in the frequency domain as a superposition of N plane waves with arbitrary orientations and unknown complex amplitudes. Specifying the pressure, the normal velocity, or the specific impedance at N locations produces a system of N linear equations with N unknowns. Then, solving for the complex amplitudes yields an approximation of the pressure and velocity fields inside the cavity. Preliminary comparisons to analytic results are presented.

**3:50**

**2pSA10. Tuned elastic shells with matched acoustic impedance and sound speed in water.** Alexey Titovich and Feruza Amirkulova (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, alexey17@eden.rutgers.edu)

Tuning of an elastic shell to match the acoustic properties of the surrounding fluid (water) is done with an internal mechanism consisting of a concentered central mass supported by an axisymmetric distribution of elastic stiffeners. The effective impedance and index are both matched at low frequencies (ka $<$ 1) by adjusting the mass and stiffness of the internal oscillator for a given shell thickness. The subsonic flexural modes of the shell, which are excited by the point attachments of the stiffeners, are ultimately suppressed in the low frequency range with a sufficiently large number of stiffeners. As a result, the scattering cross-section of the tuned shell-stiffener-mass system is negligible compared to the empty shell at frequencies below the resonance of the internal oscillator. An optimal shell thickness exists which maximizes the frequency range of water-like behavior. Furthermore, tuning the effective properties of each such shell in an array is used to achieve acoustic transparency and lensing. Inclusion of damping decreases the magnitude of the internal oscillator resonance without significantly affecting the scattered field. Several planar simulations are presented demonstrating the applications.

**4:05**

**2pSA11. Analytical scattering method of flexural waves considering the evanescent field in a thin plate.** Sungjin Cho and Junhong Park (Mech. Eng., Hanyang Univ., Haengdang-dong 17, Seongdong-gu, Seoul 133-791, South Korea, sjcho0407@hanyang.ac.kr)

This paper presents a theoretical solution on analyzing the scattering behaviors of the flexural waves by a circular obstacle medium having different material properties. In deriving the exact solution, the evanescent field together with the scattered field is considered. The modified Bessel function of the second kind is used for the evanescent field. As the evanescent wave is exponentially decaying in all directions near the edge of a circular obstacle medium when the mass density and critical angle are discontinuous, the scattering pattern without evanescent field is difficult to analyze the wave scattering phenomenon near the boundary layer. The resulting equations are solved using symbolic calculation of software MATLAB. The dependence of the scattering behaviors on the mechanical properties of the inner medium is investigated. The influence of the evanescent field on the scattering is also investigated in various configurations. The result shows that the scattering pattern has a strong dependence on the evanescent field in near field of a circular obstacle.

**4:20**

**2pSA12. Influence of cross-sectional discontinuity on the damping characteristics of viscoelastically supported rectangular plates.** Jeongwon Park, Sangkeun Ahn (Dept. of Mech. Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, jwparks@hanyang.ac.kr), Ji Woo Yoo (Res. and Development Div., Hyundai-Kia Motors, Hwaseng, Gyeonggi, South Korea), and Junhong Park (Dept. of Mech. Eng., Hanyang Univ., Seoul, South Korea)

Cross-sectional shape of a structure is one of the important design parameter associated with the vibration responses and resulting sound radiation. In this study, the influence of cross-sectional discontinuity on the vibration characteristics of plate structures was investigated. The variation of cross-sectional geometry in the rectangular plate was modeled as the change of bending stiffness for flexural wave propagation analysis. The complex translational and rotational stiffness at plate edges were used for modeling of the damping at the boundaries. The ratio between the incident and reflected waves from the boundaries was predicted for the flexural waves of different wavelengths to analyze the effect of support stiffness on the vibration damping. Using the wave propagation model, the condition of the viscoelastic boundary properties and the discontinuous flexural stiffness of plate for minimum reflection ratio, i.e., maximum dissipation of the vibration was calculated. Modal damping characteristics of the plate with and without the discontinuity in the cross-section were measured and compared to the predicted reflection ratios. The measured damping ratios on the different boundary conditions showed similar pattern with the predicted vibration energy dissipation at the viscoelastic supports.



**4:35**

**2pSA13. Modal analysis of a vibrating string via electromagnetic field excitation.** Anton A. Filyayev (Audio Arts & Acoust., Columbia College Chicago, 33 E. Congress Ave., Chicago, IL 60604, anton.filyayev@loop.colum.edu)

An electromagnetic field (EM) generating device (“EBow”) was used to excite a fixed string held in tension between two boundaries on a custom-built apparatus in the style of a monochord. A high-speed camera was used to visualize and record the vibration of the string across the length of the string for every EBow position. The data sets were analyzed to create amplitude vs time graphs. Every excitation position was found to result into a different mode of vibration that emphasized fundamental, second, third, and fourth eigenmodes. Each mode resulted in a distinct displacement pattern, which was found to follow curve fits with R<+>2<+> values ranging from 0.94 to 0.99. The recorded amplitudes also agreed with expected nodal and anti-nodal positions for every observed mode signifying the data were robust. A discrete Fourier series expansion was performed on the polynomial fits and yielded simple second-order expressions (including a damping term) which are in agreement with the expected excitation models for a fixed string.

## Session 2pSCa

## Speech Communication: Speech Technology

Yi Xu, Chair
*Univ. College London, 27 Lodge Rd., Wallington, Surrey SM6 0TZ, United Kingdom*

### *Contributed Papers*

**1:30**

**2pSCa1. Measurement of formants in synthetic vowels.** Christine H. Shadle, Hosung Nam (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu), and D. H. Whalen (Speech and Hearing Sci., CUNY Graduate Ctr., New York, NY)

The measurement of formant frequencies of vowels is among the most common measurements in speech studies, but is known to be biased by the particular fundamental frequency (F0) exciting the formants, and to be inaccurate for formants close together or for speakers using a high F0. To allow a comparison across multiple measurement techniques, vowels were synthesized using the Klatt synthesizer with known formant values. The synthetic vowels were constructed with five different F1 values and nine different F0 values; formant bandwidths, and higher formant frequencies, were constant. The F0s varied in such a way that the most intense harmonic in F1 or F2 either matched the center frequency or deviated in the range of 3–87 Hz. Manual measurements by four subjects were compared to automatic measures using the LPC Burg algorithm, LP closed-phase covariance, and spectra smoothed cepstrally or by averaging repeated DFT's. Formants were also measured from pruned reassigned spectrograms. Error patterns differ among the methods, but most tracked the frequency of the most intense harmonic; the smallest errors occur with closed-phase covariance and reassigned spectrogram. Implications for such measures on vowels in isolated words of real speech are discussed. [Work supported by NIH-NIDCD grant DC-002717.]

**1:45**

**2pSCa2. A further comparison of fundamental frequency tracking algorithms.** Hongbing Hu (Intel Corp., Binghamton, New York), Peter Guzewich, and Stephen Zahorian (Elec. and Comput. Eng., State Univ. of New York at Binghamton, PO Box 6000, Binghamton, NY 13902, zahorian@binghamton.edu)

"Yet another Algorithm for Pitch Tracking -YAAPT" was published in a 2008 JASA paper (Zahorian and Hu), with additional experimental results presented at the fall 2012 ASA meeting in Kansas City. The results presented in both the journal paper and at the fall 2012 meeting indicated that YAAPT generally has lower error rates than other widely used pitch trackers (YIN, PRAAT, and RAPT). However, even YAAPT-created pitch tracks had significant "large" errors (pitch doubling and pitch-halving) for both clean and noisy speech. Recently additional post-processing heuristics have been incorporated to reduce the incidence of these type errors—thus reducing the need for hand correcting pitch tracks for situations where extremely accurate tracks are desired. For the case of an all-voiced track, interpolation through unvoiced intervals has been improved. The updated version of YAAPT is presented along with experimental results. The experiments are conducted with multiple databases, including British English, American English, and Mandarin Chinese. For most conditions evaluated, YAAPT gives better performance than the other fundamental frequency trackers.

**2:00**

**2pSCa3. Automated assessment of English fundamental frequency contours for non-native speakers from China and India.** Keelan Evanini and Xinhao Wang (Educational Testing Service, Rosedale Rd., M.S. R-11, Princeton, NJ 08541, kevanini@ets.org)

This study investigates the F0 contours produced by non-native speakers of English from two different countries. Speakers from China ($N = 202$; L1 Mandarin) and India ($N = 230$; multiple L1s) read a paragraph out loud in the context of an assessment of non-native English speaking proficiency; in addition, a control set of native speakers ($N = 85$) read the same paragraph. The words and phonemes in all of the responses were provided with time stamps using forced alignment with the prompt text, and mean normalized F0 values were extracted for each word from each speaker. A model F0 template for the paragraph was generated by taking the mean word-level F0 value across all native speakers, and the correlation between the word-level F0 contour and this template was calculated for each non-native speaker (as in Schwanenflugel *et al.* 2004). This correlation was used as a feature for assessing F0 contours and was correlated with expert human scores of English proficiency. The results demonstrated that the feature performed somewhat better for the speakers from India ($r = 0.390$) than the speakers from China ($r = 0.326$). Furthermore, the correlations improved when function words were removed from the analysis and only content words were considered ($r = 0.396$ and $r = 0.346$).

**2:15**

**2pSCa4. Detecting voice disguise from speech variability: Analysis of three glottal and vocal tract measures.** Talal B. Amin (Elec. and Electron. Eng., Nanyang Technolog. Univ., Singapore, Singapore), James S. German (Humanities and Social Sci., Nanyang Technolog. Univ., HSS-03-46, Singapore, Singapore, jsgerman@ntu.edu.sg), and Pina Marziliano (Elec. and Electron. Eng., Nanyang Technolog. Univ., Singapore, Singapore)

The deliberate attempt by speakers to conceal their identity (voice disguise) presents a challenge for forensics and for automated speaker identification systems. Using a database of natural and disguised voices of three professional voice impersonators, we build on earlier findings (Amin *et al*., 2012) by exploring how certain glottal and vocal tract measures, including fundamental frequency (f0), glottal timing (Open Quotient), and vowel formants, are exploited to create novel voice identities. Specifically, we explored whether the amount and type of variation exhibited by impersonators can be used to develop a metric for distinguishing natural from disguised voices. As expected, variation in f0 and Open Quotient was speaker-dependent, and corresponded closely to social attributes (i.e., gender/age) of the voice identities involved. In a novel finding, the effects of voice identity on vowel formants were highly dependent on vowel category, and could not be readily characterized as global modifications to the vowel space (Bradlow *et al*.,1996). We therefore developed a no-reference objective metric for voice disguise that treats formant variability on a vowel-by-vowel basis. This metric consistently assigned high rankings to natural voices (3.3/27 on average). This correlated closely with the subjective disguisedness ratings of 18 naïve listeners, even outperforming them slightly.

**2:30**

**2pSCa5. Glottal articulations in tense vs lax phonation contrasts.** Jianjing Kuang (Linguist, Univ. of Pennsylvania, UCLA Campbell Hall 3125, Los Angeles, California 90095, kuangjj@gmail.com) and Patricia Keating (UCLA, Los Angeles, CA)

This study explores the glottal articulations of one type of phonation contrast—the tense vs lax phonation contrasts of three Yi (Loloish) languages—which is interesting because neither phonation type is very different from modal voice, and both are independent of the languages' tonal contrasts. Electroglottographic (EGG) recordings were made in the field, and traditional EGG measures showed many small but significant differences between the phonations. Tense phonation involves more overall contact and briefer but slower changes in contact. Functional Data Analysis was then applied to entire EGG pulse shapes, and the resulting first principal component was found to be mostly strongly related to the phonation contrasts, and correlated with almost all the traditional EGG measures. Unlike the traditional measures, however, this component also captures differences in abruptness of contact. Furthermore, previously-collected perceptual responses from native speakers of one of the languages correlated better with this component than with any other EGG measure or any acoustic measure. The articulatory differences between these tense and lax phonations, involving glottal aperture and how glottal closure is made, are not extreme, but apparently they are consistent enough, and perceptually robust enough, to support this linguistic contrast. [Work supported by NSF.]

**2:45–3:00 General Discussion**

**3:00–3:30 Break**

**3:30**

**2pSCa6. The effect of non-linear dimension reduction on Gabor filter bank feature space.** Hitesh A. Gupta, Anirudh Raju, and Abeer Alwan (Elec. Eng., Univ. of California Los Angeles, 550 Veteran Ave., Apt. 102, Los Angeles, CA 90024, hiteshag@ucla.edu)

In this paper, we modify the Gabor feature extraction process, while applying the Gabor filters on the power-normalized spectrum and concatenating with power normalized cepstrum coefficients (PNCC), for noise robust large vocabulary continuous speech recognition. In Chang *et al.*, ICASSP (2013), a similar Gabor filter bank (GBFB) feature set with multilayer perceptron (MLP) processing (to reduce the feature dimension) has been used with mel frequency cepstrum coefficients showing improvements on Aurora-2 and renoised Wall Street Journal corpora. On a subset of the Aurora-4 database (only male), our method has shown promising results (when using PCA) being 7.9% better than 39-dimensional PNCC features. But, the GBFB features are a rich representation of the speech spectrogram (as an overcomplete basis), and an appropriate dimension reduction/manifold learning technique is the key to generalizing these features for the large vocabulary task. Hence, we propose the use of Laplacian Eigenmaps to obtain a reduced manifold of 13 dimension (from a 564-dimensional GBFB feature set) for the training dataset with a MLP being used to learn the mapping so that the same can be applied to out-of-sample points, i.e., the test dataset. The reduced GBFB features are then concatenated with the 26-dimension PNCC plus acceleration coefficients. This technique should lead to better accuracies as speech lies on a non-linear manifold rather than a linear feature space. [This project was supported in part by DARPA.]

**3:45**

**2pSCa7. Language material for English audiovisual speech recognition system development.** Andrzej Czyzewski (Multimedia Systems Dept., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl), Tomasz Ciszewski, Dorota Majewicz (Faculty of Philology, Univ. of Gdansk, Gdansk, Poland), and Bozena Kostek (Multimedia Systems Dept., Gdansk Univ. of Technol., Gdansk, Poland)

The bi-modal speech recognition system requires a 2-sample language input for training and for testing algorithms which precisely depicts natural English speech. For the purposes of the audio-visual recordings, a training data base of 264 sentences (1730 words without repetitions; 5685 sounds) has been created. The language sample reflects vowel and consonant frequencies in natural speech. The recording material reflects both the lexical word frequencies and casual speech sound frequencies in the BNC corpus of approx. 100m words. The semantically and syntactically congruent sentences mirror the 100m-word corpus frequencies. The absolute deviation from source sound frequencies is.09% and individual vowel deviation is reduced to a level between 0.0006% (min.) and 0.009% (max.). The absolute consonant deviation is 0.006% and oscillates between 0.00002% (min.) and 0.012% (max.). Similar convergence is achieved in the language sample for testing algorithms (29 sentences; 599 sounds). The post-recording analysis involves the examination of particular articulatory settings which aid visual recognition as well as co-articulatory processes which may affect the acoustic characteristics of individual sounds. Results of bi-modal speech elements recognition employing the language material are included in the paper.

**4:00**

**2pSCa8. Characteristics of automatic and human speech recognition processes.** Mark VanDam (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu) and Noah H. Silbert (Commun. Sci. & Disord., Univ. of Cincinnati, Cincinnati, OH)

In a previous report [VanDam and Silbert (2013) *POMA* **19**, 060006], we investigated performance of a commercially available automatic speech recognition (ASR) system [LENA Research Foundation, Boulder, CO] on acoustic recordings from family speech in naturalistic environments. We found that the ASR more accurately labeled children over adults and fathers over mothers, and human judge labels included substantial individual variation. The present work extends previous work by investigating the possible sources for both machine- and human labeling decisions. Classification tree models were fit to several acoustic variables for machine- and human labels of *CHILD*, *MOTHER*, and *FATHER*. Results suggest that (a) fundamental frequency ($f_0$) and duration measures influenced label assignment for both machine and human classifications, (b) the error of the fitted models is lower for the machine labeling procedure than for human judges, (c) machine- and human decision processes use the acoustic criteria (i.e., $f_0$ and duration) differently, and (d) $f_0$ is more important than duration for all labelers. Results may have implications for improving implementation and interpretation of ASR techniques, especially as they are useful for understanding child language applications and very large, naturalistic datasets that demand unsupervised ASR techniques.

**4:15**

**2pSCa9. Crying for help: The Frye hearing and forensic acoustic analyses in State of Florida vs George Zimmerman.** Al Yonovitz (The Univ. of Montana, Dept. of Communicative Sci. and Disord., Missoula, MT 59812, al.yonovitz@umontana.edu), Herbert Joe (Yonovitz and Joe, LLP, Irvine, CA), and Joshua Yonovitz (Yonovitz and Joe, LLP, Missoula, Montana)

Neighborhood watch volunteer George Zimmerman observed suspicious activity and called police. In minutes, he and Trayvon Martin were in an altercation when Mr. Zimmerman shot and killed Trayvon Martin and claimed self defense. The State's audio experts evaluated the 9-1-1 audio recordings to determine who spoke which background phrases, if any. A Frye hearing is where the court determines the reliability and admissibility of an expert's opinion by determining whether an expert's methodologies are generally accepted within the relevant scientific community. The judge in this case ruled that the State's two audio experts would not be permitted to testify during the trial. While the judge found the "aural perception and spectral analysis... are sufficiently established to have gained general acceptance within the scientific community," she took exception to how they were applied in this case. This case provided a venue and forum for opinions on voice analysis, methodologies, standards, and the quality required of evidentiary audio for determination of speaker identification and elimination. Positions taken by witnesses, a critique of the Frye Hearing and the scientific basis for witness and legal conclusions will be discussed.

**4:30**

**2pSCa10. Intensity slopes as robust measure for distinguishing glottalic vs pulmonic stop initiation.** Sven Grawunder (Dept. of Linguist, Max Planck Inst. for Evolutionary Anthropology, Deutscher Platz 6, Leipzig 04103, Germany, grawunder@eva.mpg.de)

A novel cross-linguistically robust measure is introduced for the linguistically relevant distinction of pulmonic vs glottalic (ejective) stops.

We propose to parametrize the abruptness of the following vowel onset by using intensity slope (RMS-trajectory) at the voicing onset. This measure was previously discussed only for voicing distinction of pulmonic stops (Harrington, 2012). The dependencies on vowel quality of the following vowel, voice onset time (VOT), phonetic prominence and speaking rate are investigated on a small-scale sample of two speakers from Avar (Nakh-Dagestanian), Ingush (Nakh-Dagestanian), and Georgian (Kartvelian). The results demonstrate robust significant differences of intensity slopes between pulmonic and glottalic stops, the latter showing the steepest slopes/the most abrupt onsets. In some cases of prevoiced stops, the intensity slopes allow even a tripartite distinction of (pre-)voiced, voiceless (aspirated), and ejective stops. However, in order to countervail the influence of vowel specific onset characteristics (e.g., degree of lip rounding) or the influence of VOT, the sample must be controlled for vowel quality and place of articulation. And, since higher speaking rates and less prominent syllables involve higher rates of laryngealized (creaky) vowel onsets, a breakdown of the abruptness distinction is observed under these conditions.

**4:45–5:00 General Discussion**

TUESDAY AFTERNOON, 3 DECEMBER 2013 PLAZA A, 1:00 P.M. TO 5:00 P.M.

## Session 2pSCb

## Speech Communication: Speech Perception II (Poster Session)

Meghan Sumner, Chair
*Dept. of Linguist., Stanford Univ., Margaret Jacks Hall, Bldg. 460, Stanford, CA 94305-2150*

### *Contributed Papers*

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

**2pSCb1. Perception of Scottish Gaelic alternating (leniting) consonants.** Natasha L. Warner (Dept. of Linguist, Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), Ian Clayton (Dept. of English, Boise State Univ., Boise City, ID), Andrew Carnie, Muriel Fisher, Dan Brenner, Michael Hammond, Diana Archangeli, and Adam Ussishkin (Linguist, Univ. of Arizona, Tucson, AZ)

Scottish Gaelic, an endangered Celtic language, demonstrates alternations in word-initial consonants, as in “pòg” [$p^h$ok] ’kisses’ vs “phòg” [fok] ’kissed.’ This process, called lenition, leads to apparent neutralizations of Gaelic segments, for example of the [f] of “ phòg” with [f] of “foghlam” [foɫəm] “education,” which is not caused by lenition. A perception experiment can show whether listeners hear any residual difference between lenited segments (e.g., [f]<-[$p^h$]) and the phonetically similar segments ([f]<-[f]). This project used a gating study to investigate when in the word listeners determine which type of sound they are hearing. Preliminary results from 17 native Gaelic listeners indicate that listeners cannot distinguish lenited from phonetically matched consonants (e.g., the two types of [f]) from cues in the consonant itself, but can distinguish both from the unlenited phonologically matched consonants (e.g., [$p^h$]) very accurately. Listeners become able to distinguish lenited from phonetically matched segments (the two types of [f]) during either the following vowel or the segment after that, depending on what coarticulatory cues with the latter parts of the word are available. Thus, listeners need enough acoustic information to provide lexical disambiguation in order to determine the morphological source of lenited sounds.

**2pSCb2. An online investigation of compensation for coarticulation in Mandarin learners of English.** David Li (Linguist, Univ. of Southern California, 2133 Sun Ridge Dr., Chino Hills, CA 91709, li.david.c@gmail.com) and Elsi Kaiser (Linguist., Univ. of Southern California, Los Angeles, CA)

Phonological variation can cause lexical ambiguity: The word “run” may sound like “rum” in sentences like “A quick run picks you up” (Gaskell and Marslen-Wilson, 2001) because /n/ is assimilated to /m/ due to the influence of /p/. Previous studies have shown that listeners compensate for coarticulation, perceiving acoustically identical sounds differently in different contexts (e.g., Mann, 1980). We investigated whether native Mandarin speakers with English as a second language (L2) compensate for assimilation in the same manner/to the same extent as native English speakers (L1). Given that there are no Mandarin words with the nasal coda [m], do Mandarin learners of English compensate for phonological variation not present in their L1? We conducted a visual-world eye-tracking study—using English stimuli—to investigate compensation-for-assimilation in L1 English and L1 Mandarin speakers. Identical acoustic sequences were embedded in carrier sentences with fast vs slow speech rates to test interpretation of potentially ambiguous words (cf. rum/run) by L1 and L2 speakers. Eye-movements reveal real-time differences in how L1 and L2 speakers interpret coarticulated speech, with L1 speakers showing signs of rapid compensatory processes. We also discuss how listeners’ interpretation is influenced by other acoustic cues in coarticulated speech.

**2pSCb3. Classification of affricate burst place in consonant-vowel contexts in English.** Jung-Won Lee, Hong-Goo Kang (Elec. and Electron. Eng., Yonsei Univ., 134 Shinchon-dong, Seodaemun-gu, Seoul 120-749, Korea, Republic of, jaesuk2002@dsp.yonsei.ac.kr), and Jeung-Yoon Choi (Res. Lab. of Electron., Massachusetts Inst. of Technol., Cambridge, MA)

This study investigates characteristics of affricate burst place of articulation compared with the bursts for the three places of articulation for stops (labial, alveolar, and velar) in English. The data comprise consonant-vowel tokens in the TIMIT corpus. To assess which stop place of articulation may be used to simultaneously model affricate bursts, Jensen-Shannon divergence measures are found for probability distributions of acoustic-phonetic features. In addition, we conduct classification experiments using combinations of acoustic-phonetic features and Mel-frequency cepstral coefficients (MFCCs), to see how well affricate burst place is classified using models for the three stop places. The experimental results show that although affricate place is similar to the alveolar place of articulation for stops, a separate post-alveolar place for affricate burst provides a better model. The results suggest that a separate affricate place model will be useful in a feature-based speech recognition system that explicitly detects place of articulation for consonants.

**2pSCb4. Effect of consonantal place of articulation on the perception of phonetic voicing in plosives.** Viktor Kharlamov and Anna Loukianova (Linguist, Univ. of Arizona, Box 210028, Tucson, AZ 85721, kharlamov@email.arizona.edu)

For plosive consonants, production of phonetic voicing often varies across consonantal places of articulation. Anterior stops (e.g., /b/, /d/) tend to show more glottal pulsing compared to the plosives produced towards the back of the oral cavity (e.g., /g/), which is traditionally attributed to the smaller volume of air passing through the glottis during the closure stage of posterior stops. The current work demonstrates that a similar effect of consonantal place of articulation also exists in perception. Results of a series of identification experiments show that English listeners ($n = 60$) are more sensitive to glottal pulsing in word-final dorsals than labials or coronals. The effect is especially robust when, within each experimental block, listeners are exposed to variability in consonantal place of articulation (i.e., when labial, coronal, and dorsal plosives are blocked together) but not to variability in phonetic vowel duration (i.e., when a given block contains only short or long vowels). This suggests that glottal pulsing is more perceptually salient in posterior plosives, for which voicing is less expected in production, and that the sensitivity to voicing in dorsals is stronger when listeners can do an online comparison of voicing cues across different places of articulation.

**2pSCb5. Discrimination between fricatives and affricates pronounced by Japanese native speakers at various speaking rates.** Kimiko Yamakawa and Shigeaki Amano (Human Informatics, Aichi Shukutoku Univ., 9 Katahira, Nagakute, Aichi 4801197, Japan, jin@asu.aasa.ac.jp)

Fricatives [s] and affricates [ts] uttered at a normal speaking rate are successfully discriminated with two variables: a rise part duration and a steady + decay part duration [Yamakawa *et al.*, Acoust. Sci. Tech. **33**(3), 154–159 (2012)]. This study examined whether [s] and [ts] uttered at various speaking rates are also well discriminated with these variables. Discriminant analyses with the two variables were performed on [s] and [ts] in word-initial position in a carrier sentence pronounced by eight native Japanese speakers at fast, normal, and slow speaking rates. Discriminant error rates were low for the fast (9.8%, $n = 512$), normal (5.7%, $n = 512$), and slow (13.1%, $n = 512$) speaking rates. However, the error rate was high (22.0%, $n = 1536$) when the three speaking rates were analyzed together. It decreased to 15.7% when the two variables were normalized with an averaged mora duration of the carrier sentence. These results suggest that [s] and [ts] can be discriminated at various speaking rates with a normalized rise part duration and normalized steady + decay part duration. [This work was supported by JSPS KAKENHI Grant Numbers 22720173, 24652087, and 25284080, and by Aichi Shukutoku University Special Research Grant 2011-2012 and Corporate Research Grant 2013-2014.]

**2pSCb6. Relationship of training of pitch-pattern reconstruction ability to speech perception in normal-hearing young adults.** Kristen M. Cortese (Commun. Disord. & Sci., Rush Univ., 1001 West Cypress Dr., Arlington Heights, IL 60005, kristen_cortese@rush.edu), Stanley Sheft, Valeriy Shafiro, and Derek J. Stiles (Commun. Disord. & Sci., Rush Univ., Chicago, IL)

Previous research has indicated that processing of pitch changes is related to speech perception. Current work investigated the relationship of pitch-pattern training to speech perception in young normal-hearing adults. The training protocol was based on an interactive pattern-reconstruction task in which listeners assembled four or five tones (frequency: 400–1750 Hz, duration: 75–600 ms) to match a random target sequence. Control and experimental groups, of 13 subjects each, were tested. The study included a pretest, three training sessions (experimental group only), and a posttest. Training consisted of three 40-min sessions of interactive pitch-pattern reconstruction. In the pre- and posttest, listeners also reconstructed patterns of sinewave speech, and completed measures of speech perception which included sinewave speech and speech-in-noise tasks. Despite training with only tonal patterns, results showed significant improvement in the ability to reconstruct patterns for both tonal and sinewave-speech stimuli. Significant improvements were obtained for intelligibility of sinewave speech between pre- and posttest, which were greater for the experimental than control group. A greater relationship between results from the intelligibility and pattern-reconstruction conditions post training was also found. Overall, results suggest that the training protocol may benefit speech perception, especially in conditions of degraded speech. [Work supported by NIDCD.]



**2pSCb7. Comprehending speech at artificially induced rates.** Lucia da Silva (Linguist, Univ. of Br. Columbia, 6335 Thunderbird Crescent Box 137, Vancouver, BC V6T2G9, Canada, lucia@alumni.ubc.ca), Adriano V. Barbosa (Dept. of Electronics, Federal Univ. of Minas Gerais, Belo Horizonte, Brazil), and Eric Vatikiotis-Bateson (Linguist., Univ. of Br. Columbia, Vancouver, BC, Canada)

This study confirms and extends Fairbanks, Guttman and Miron's study (1957), where comprehension of time-compressed (2:1), recorded speech was improved when played back twice. Specifically, we examine the perception of pitch corrected and non-pitch corrected compressed speech when presented twice in succession and twice with delays of minutes between stimulus presentations. Additionally, we investigate whether visual speech information aids or hinders comprehension by comparing fast rate effects when presented acoustically and audiovisually. A motivation for this study is the increasing dependency on web-based multimedia recordings for knowledge transfer.

**2pSCb8. Perceptual interactions among components of a spectral-domain voice source model.** Marc Garellek (Linguist. Dept., UCSD, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, mgarellek@ucsd.edu), Robin A. Samlan, Jody Kreiman, and Bruce Gerratt (Dept. of Head and Neck Surgery, UCLA, Los Angeles, CA)

A psychoacoustic model of the source spectrum has been proposed in which source contributions to overall voice quality can be quantified by four spectral slope components: H1-H2 (the amplitude difference between the first and second harmonics), H2-H4, H4-2000 Hz (i.e., the harmonic nearest to 2000 Hz), and 2000–5000 Hz. The natural variability of these components has been described, along with the just noticeable differences (JNDs) for each component. The goals of this study are to identify how perceptual sensitivity to each component slope varies as a function of the adjacent slope(s). The JNDs were obtained for stimuli based on synthetic copies of one female voice. The stimuli were manipulated so that spectral slope varied in 0.5 dB increments (1 dB for 2000–5000 Hz) for a particular component at "high" and "low" values of the adjacent slopes. Thirty-three listeners completed an adaptive up-down paradigm. Preliminary results suggest that sensitivity to a particular spectral component varies as a function of some adjacent components more than the other. Furthermore, sensitivity varies based on whether the adjacent component has a high or low slope. These interactions among spectral components will be interpreted with respect to the variability in spectral configuration observed in 144 voices.

**2pSCb9. Lexical segmentation of speech from energy above 5 kHz.** A. Davi Vitela (Univ. of Arizona, 1131 E. 2nd St., Tucson, AZ 85721, adv1@email.arizona.edu), Brian B. Monson (DUKE-NUS, Singapore, Singapore), and Andrew J. Lotto (Univ. of Arizona, Tucson, AZ)

Research in the field of speech perception has traditionally focused on the acoustic information or cues present in the frequency region below 5 kHz, thus ignoring high frequency energy (HFE). A recent series of studies, however, demonstrated that listeners could determine the mode of production (speech or singing) and further, could identify what was being spoken or sung from the HFE alone and with a speech-shaped masking noise in the lower frequencies. This begs the question as to what types of information listeners are extracting to guide their perception. The current study examined the ability of listeners to transcribe short semantically-unpredictable but syntactically-well-formed spoken phrases that were high-pass filtered at 5.6 kHz. Of particular interest was the ability of some listeners to correctly determine the placement of word boundaries even without the availability of low-frequency information. These findings add to a growing literature on the linguistically relevant information present in higher frequency regions. Results will be framed within current theories of acoustic signatures to lexical segmentation. [Work supported by NIH-NIDCD.]

**2pSCb10. Prosodic disambiguation of Korean relative clause attachments.** Younah Chung (Linguist., UCSD, UCSD Linguist Dept., 9500 Gilman Dr., La Jolla, CA 92093-0108, yachung@ucsd.edu) and Chongdok Kim (Inst. of Lang. and Information Studies, Yonsei Univ., Seoul, South Korea)

Korean relative clauses do not have to be adjacent to the substantives they modify; genitive noun phrases can interpose between the relative clause and the substantive. Thus, the syntactic analysis of a given relative clause is inherently ambiguous, such that the relative clause can modify either the head noun of a genitive noun phrase or the complex noun phrase as a whole. However, in spoken language, one rarely finds this type of ambiguity even though the relative clause could modify either of the two nouns. This suggests that speakers use prosody to convey intended meaning, with different prosodic cues associated with different syntactic boundaries, thus, providing information necessary for disambiguation. We examined the clarification of surface structure ambiguity in Korean relative clause attachments via prosody. The survey confirmed that native speakers think relative clauses are surface ambiguous. The acoustical analysis revealed that prosodic disambiguation consists of pause insertion, pitch raising, and final lengthening at the syntactic boundary. Regarding the relative importance of different prosodic measures used by speakers, we discovered that pitch raising was a more powerful measure than final lengthening in prosodic disambiguation, and that pausing played the least important role in the parsing of an ambiguous relative clause.

**2pSCb11. The effect of phonetic orthography on the perception of Mandarin syllables.** Yu-Jung Lin, Chung-Lin Yang (Linguist., Indiana Univ., 800 N Union St. Apt. 405, Bloomington, IN 47408-2230, lin41@indiana.edu), and Chien-Jer Charles Lin (East Asian Lang. and Cultures, Indiana Univ., Bloomington, IN)

It has been found that vowels have perceptual advantage over tones (e.g., Ye and Connie, 1999) and that phonetic orthographic information can be activated in speech perception (e.g., Frauenfelder, Segui, and Dijkstra, 1990). In the present study, we used a tone-vowel detection task where Zhuyin (Taiwanese) and Pinyin users (Mainland Chinese) were asked to monitor Mandarin syllables containing [i4] to test how tonal and vowel information are processed in syllable monitoring, and to examine if Zhuyin and Pinyin users' responses are affected by their phonetic orthographic systems. The results demonstrated that in both groups, vowel-mismatched syllables (e.g., bu4) were rejected faster than tone-mismatched syllables (e.g., bi2). Moreover, the RT of rejecting vowel-mismatched syllables and double-mismatched syllables (e.g., bu2) were similar, indicating that as long as vowel was mismatched, tonal information caused little difference in RT when rejecting the mismatched syllables. This finding was consistent with Ye and Connine (1999). Furthermore, both groups resort to their phonetic orthography in simple syllables (e.g., bi4, zhi4). However, in complex syllables (e.g., biao4), different patterns were observed between the two groups, possibly due to the factors such as how tones are marked in their phonetic orthography or how they learned the orthography.

**2pSCb12. Perceiving politeness from speech acoustics alone: A cross-linguistic study on Korean and English.** Bodo Winter (Cognit. and Information Sci., Univ. of California, Merced, 3681 San Jose Ave., Apt. 4, Merced, CA 95348, bodo@bodowinter.com), Lucien Brown, Kaori Idemaru (East Asian Lang. & Literatures, Univ. of Oregon, Eugene, OR), and Sven Grawunder (Linguist., Max Planck Inst. for Evolutionary Anthropology, Leipzig, Germany)

Politeness is a crucial aspect of everyday speech communication; however, there are to date only few acoustic studies on this topic. Winter and Grawunder (2012) showed that for Korean speakers, politeness is reflected in pitch, intensity, voice quality and speaking rate. Here, we extend this production study by investigating whether Korean and English listeners can perceive the intended politeness of short Korean utterances based on speech acoustics alone. In two experiments with a total of 47 English and 30 Korean listeners, we found that both groups can detect the intended politeness purely based on the phonetic qualities of speech. In one experiment, accuracy was low (Korean: 58%, English: 53%) because speakers heard multiple voices in a randomized fashion, not allowing them to familiarize with any particular voice. In a design that was blocked by speaker voice, accuracy was higher (Korean: 70%, English: 58%), showing that vocal politeness can be used as a cue when the voice is known. This shows that politeness is not only expressed by honorific lexical forms commonly employed in Korean, but also by speech acoustics. It is remarkable that English speakers performed above chance at all, pointing to cross-linguistic regularities in how politeness is expressed vocally.

**2pSCb13. The role of vowels in determining a male speaker's sexual orientation.** Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research (Tracy and Satariano, 2011) found that listeners primarily use the vowel in a monosyllabic word to determine the speaker's sexual orientation. Given this result, if listeners were presented utterances that contained multiple vowels, then the accuracy of their sexual orientation ratings should increase. In Experiment 1, listeners were presented with trisyllabic words ("authentic"). They heard the first syllable (/ə/), the first two syllables (/əθɛn/), and the entire word (/əθɛntɪk/). Sexual orientation ratings were the most accurate for the entire word and the least accurate for the first syllable. These results were replicated in experiment 2, which used only bisyllabic words. An items analysis from both experiments revealed that a syllable that contained more phones resulted in higher accuracy ratings compared with a syllable that contained fewer phones. For example, listeners' sexual orientation ratings were more accurate when they heard the first syllable of "lavender" compared to the first syllable of "authentic." These findings suggest that when the number of vowels is kept constant, listeners' responses are more accurate if the utterance contains additional phones.

**2pSCb14. Acoustic features mediating height estimation from human speech.** John Morton (Psych., Washington Univ., 2309 Laurenwood Dr., Chesterfield, MO 63017, jrmhvc333@yahoo.com), Mitchell Sommers (Psych., Washington Univ., St. Louis, MO), Steven Lulich (Linguist, Indiana Univ., Bloomington, IN), Abeer Alwan, and Harish Arsikere (Eng., UCLA, Los Angeles, CA)

The current experiment was aimed at providing the first direct evidence regarding the role of subglottal resonances in height discrimination. Past research has investigated whether or not listeners are able to discriminate which of two talkers is taller (Rendall *et al*., 2007), but has not established what parameters of the acoustic speech signal listeners use to distinguish speaker height. In the current study, we examined the role of subglottal resonances in height discrimination. Subglottal resonances generally refer to resonances of the lower airways starting in the lungs and terminating at the glottis. Subglottal resonances would be a good candidate for use in height estimation because they remain relatively stable across vowels and are relatively unaffected by other factors influencing the supra-laryngeal vocal tract; instead they are influenced nearly exclusively by the lower airway acoustic

properties. Listeners participated in two tasks, the first task was a two-alternative forced choice (2AFC) height discrimination test, in which listeners heard sentences produced by two talkers of the same gender and were asked to determine which of the two was the tallest. The second task involved listening to five talkers (all same gender) sentences, and then ranking those individuals from tallest to shortest. Findings indicate that listeners are able to discriminate and rank speakers heights better than chance, and the role of subglottal resonances will be reported.

**2pSCb15. Lexically biased perceptual adaptation.** Michael McAuliffe (Linguist., Univ. of Br. Columbia, Totem Field Studios, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, mcauliff@interchange.ubc.ca)

Listeners are able to rapidly shift their acoustic-phonetic categories in the face of speaker-specific variation. However, an open question is how top-down expectations can influence this perceptual adaptation. Under Adaptive Resonance Theory [Grossberg (2003)], if an acoustic token provides initial activation of a linguistic category, top-down expectations about that category can boost the resonance and lead to activation of the category even with an ambiguous token. Ambiguous tokens in positions with stronger expectations are hypothesized to lead to less perceptual adaptation than in positions with lower expectations. To test this hypothesis, two groups of participants will be exposed to 10 ambiguous s-ʃ sounds during a lexical decision exposure task. The groups differ in where in the word these sounds will occur, either in the onset or coda of CVC words. Lexical bias, as a form of top-down expectations, is stronger for ambiguous coda sounds [Pitt and Szostak (2012)]. I predict that in a s-ʃ categorization task following exposure, the coda group will differ less than the onset group from a control group that only completes the categorization task, indicative of less perceptual adaptation.

**2pSCb16. Degraded word recognition in isolation vs a carrier phrase.** Kathy M. Carbonell and Andrew J. Lotto (Speech Lang. & Hearing Sci., Univ. of Arizona, 1131 E 2nd St., Tucson, AZ 85721, kathyc@email.arizona.edu)

Recognizing a spoken word presented in isolation is a markedly different task from recognizing a word in a carrier phrase. The presence of a carrier phrase provides additional challenges such as lexical segmentation but also provides additional information relevant to word recognition such as speaking rate and talker-specific spectral characteristics. The current set of studies is part of an attempt to determine how target word recognition differs in isolation versus in a carrier phrase. In an initial experiment, a set of 220 spoken CVC words were noise-vocoded (6 channel) and presented to listeners either in isolation or following a noise-vocoded carrier phrase—"The next word on the list is…" The target words were transcribed in each condition and scored for initial consonant accuracy and overall word accuracy. Despite the lack of semantic or syntactic information provided by the carrier phrase accuracy for both word and consonant recognition were much higher in the carrier phrase context. A second experiment, used different talkers for the carrier phrase and target word. A smaller but significant benefit over isolation was present for these mixed talker stimuli suggesting that the benefits of the carrier phrase include, but are not limited to, talker-specific information.

**2pSCb17. Differences in the recognition of careful and casual speech.** Meghan Sumner, Jeremy Calder, Annette D'Onofrio (Dept. of Linguist, Stanford Univ., Margaret Jacks Hall, Bldg. 460, Stanford, CA 94305, sumner@stanford.edu), Kevin B. McGowan (Dept. of Linguist, Stanford Univ., Stanford, Michigan), and Teresa Pratt (Dept. of Linguist, Stanford Univ., Stanford, CA)

Previous work in spoken word recognition and speech perception has shown two seemingly conflicting patterns. While some studies have shown a processing benefit for more frequent word variants (i.e., in a casual speech mode), others have found a benefit for more canonical word forms (i.e., in a careful speech mode). This study aims to reconcile these findings, proposing that different types of processing apply to each speech mode –top-down processing for casual speech, and bottom-up for careful speech. Listeners in an auditory priming task heard natural (non-spliced) sentences spoken in either a careful or casual speech mode. The final word of the auditory prime was either semantically predictable from the preceding sentence context or unpredictable. After the audio prime, listeners responded in a lexical decision task to a visual probe: either the final word heard in the prime, an unrelated word, or a nonword. Preliminary results suggest that, regardless of speech style, reaction times are faster for related targets in the semantically predictable conditions than for unrelated targets. Crucially, responses to the target word in the careful condition are delayed compared to casual speech for semantically unpredictable sentences. The implications for the apparent paradox in previous results will be discussed.

**2pSCb18. Use of acoustic cues in the perception of complex syllable structure.** Goun Lee (Linguist., Univ. of Kansas, 1541 Lilac Ln., Blake Hall, Rm. 427, Lawrence, KS 66045-3129, cconni@ku.edu)

This study examines second/foreign language learners' (L2ers') use of acoustic cues in the perception of syllable structure (onset clusters) that does not exist in the native language (L1, here, Korean). It investigates whether the target-like perception of syllable structure is facilitated by VOT ([stɪn] vs [sətɪn]) and partial devoicing of the liquid ([plim] vs [pəlim]) as compared to when either cue is absent ([blint] vs [bəlint]). This study also examined whether production errors can be predicted by perception errors. Twelve Korean-speaking English L2ers and 12 native English speakers completed AXB and production tasks in which they listened to and produced nonce words in the above three conditions. The AXB results showed significant effects of L1 and condition, but no interaction between the two. Participants were most accurate on the condition with VOT (English: 98%, Korean: 85%), followed by partial devoicing of the liquid (English: 96%, Korean: 79%) and without either cue (English: 94%, Korean: 74%). L2ers' perception and production patterned alike, but no correlation was found between the two. These findings suggest that acoustic cues like VOT and partial devoicing of the liquid facilitate L2ers' perception of new syllable structure, but do not play a direct role in its production.



**2pSCb19. The perception of English [h] and [ɹ] by Brazilian Portuguese speakers.** Denise M. Osborne (Univ. of Arizona, 814 E. 9th St. Apt 7, Tucson, AZ 85719, dmdcame@hotmail.com)

Brazilian Portuguese (BP) learners of English have difficulties in differentiating between initial English [h] and [ɹ] (Zimmer, Silveira, and Alvers, 2009). This study investigates how speakers who have BP as their L1 and English as their L2 perceive the phonetic distance of English [h] and [ɹ], and how they and monolingual BP speakers map these phonemes onto BP sound categories. 32 native BP learners of English participated in three consecutive experiments: AXB Discrimination, Identification, and Assimilation Tests. In addition, 18 monolingual BP speakers participated in the same Assimilation test. Significant effects in ANOVAs on the three experiments support the results. Beginners and intermediates were able to hear the distinction acoustically. Only intermediates, however, used the distinction to identify English words. Both monolingual BP speakers and intermediates assimilate English [h] primary to BP double <rr> (reflecting the dialectal and allophonic variation of the BP rhotic sounds). Beginners showed a failure to assimilate L2 sounds to their L1 BP categories. Comparison across these experiments shows that, at some stages of learning, BP speakers experience difficulty in evaluating these L2 sounds relative to their L1 inventory and in using the L2 distinction at the lexical level.

**2pSCb20. Effects of hearing loss and linguistic context for phoneme and word recognition in noise.** Adam Svec, Benjamin Munson, and Peggy B. Nelson (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, svecx002@umn.edu)

This project measured whole-word and phoneme recognition in normally hearing (NH) and hearing-impaired (HI) participants listening in steady-state noise. Stimuli were modeled after those used in Olsen *et al.* *[EarHear 1997]* including 10-word lists of consonant-vowel-consonant (CVC) monosyllables containing 10 vowels and 20 consonants. Words were presented in isolation and in contextually correct sentences. Speech-shaped noise was presented at a range of signal-to-noise ratios, from −9 to +3 dB SNR for NH listeners, and from −6 to +6 dB SNR for HI listeners. Results suggested that HI listeners needed an increased proportion of components ($p_p$) to recognize a whole word ($p_w$) relative to their NH counterparts (e.g., a higher $j$ value) when stimuli were presented at unfavorable SNRs. For -3 and 0 dB SNR conditions, j factors for NH listeners (0 dB SNR: $j = 2.17$) were consistently lower than those for HI listeners (0 dB SNR: $j = 2.52$). However, for the 3 dB SNR conditions, j factors were higher for NH listeners ($j = 3.06$) than those for HI listeners ($j = 2.58$). This suggests an unexpected interaction between hearing acuity, background noise, and the recognition of phonemes versus whole words. [Research supported by DC008306 to Peggy Nelson.]

**2pSCb21. Sociolinguistic perceptions of Californian /æ/ backing.**
Annette D'Onofrio (Linguist., Stanford Univ., 450 Serra Mall, Bldg. 460, Stanford, CA 94305, annetted@stanford.edu)

This paper investigates perceptions of a feature of the "California Vowel Shift" (Eckert, 2004): the backing of the TRAP vowel. The present study probes whether or not Californian and non-Californian listeners exhibit knowledge of TRAP backing's dialectal patterning in perception. An American English speaker's TRAP vowel was manipulated to create a 9-step continuum from /a/ to /æ/ in a sentence frame. 144 native U.S. listeners heard one of these steps, accompanied by orthographic information about the word spoken (e.g., the word was explicitly given as "blocked" or "blacked"). Listeners provided social categorizations of the speaker. Listeners were more likely to categorize the speaker as Californian when presented with TRAP orthography (e.g., "blacked") paired with a token on the continuum that was closer to /a/. 60 listeners participated in a phoneme categorization task using the same manipulated stimuli. When listeners were told that a speaker was from California, they were more likely to place the boundary between "blacked" and "blocked" closer to the /a/ end of the continuum than if told the speaker was from elsewhere. Together, these experiments indicate that listeners exhibit awareness of TRAP-backing's dialectal association with California, the top-down knowledge of which can affect lexical categorization.

**2pSCb22. Detecting speaker change in background audio streams.**
Hilary Toh, Pei Xuan Lee (Victoria Junior College, 23 Li Hwan View, Singapore, Singapore 556913, Singapore, toh.si.yin.hilary.2011@vjc.sg), Boon Pang Lim, and Nancy F. Chen (Human Lang. Technol., Inst. for Infocomm Res., Singapore, Singapore)

The cocktail party effect refers to the ability of humans to selectively focus on one speech stream among multiple human-generated background conversations and noise sources. This ability is influenced by various factors including directionality, speaking rates, different speech accents and speaking styles. It is displayed most clearly as a binaural effect, requiring listeners to be able to identify and differentiate between multiple sound sources simultaneously. Past studies [Cherry, J. Acoust. Soc. Am. **25**(5), 975–979 (1953)], have shown that listeners can detect a change of speaker voice from male to female in unattended audio streams, while focusing their attention on repeating speech in an attended audio stream. Our work further investigates this by focusing on speaker change across the same gender: a more difficult task given a smaller variance in pitch across speakers. Experiments were conducted on 24 listeners and findings suggest that they are able to recognize speaker changes of the same gender. Two factor ANOVA shows an interaction between listener and speaker gender ($F(1, 22) = 7.688$, $p = 0.01$), suggesting listeners perform better when detecting speaker changes within their own gender.

**2pSCb23. A word to the eyes: Visual cues benefit lexical segmentation in noise.**
Kate Helms Tillery, Sarah J. Cook, Rene L. Utianski, Julie M. Liss, and Michael F. Dorman (Dept. of Speech and Hearing Sci., Arizona State Univ., Coor Hall 2211, P.O. Box 870102, Tempe, AZ 85287-0102, kate.helms-tillery@asu.edu)

In English, most word-initial syllables are stressed. Listeners use these syllable strength cues to identify word boundaries in degraded acoustic conditions. This is evident in the types of lexical boundary errors they make when tasked with parsing a continuous stream of degraded speech: word boundaries are more often inserted before strong syllables than before weak syllables. Listeners also use visual cues from the talker's face to glean both phonemic and prosodic speech information in degraded listening conditions. While the benefits of lip-reading have received much attention, the role of visual cues in lexical segmentation remains largely unexplored. The present study examined the effect of auditory-visual cues on lexical boundary decisions. Normal-hearing listeners identified target phrases degraded by multi-talker babble. Responses in auditory-only and auditory-visual conditions were analyzed for percent words correct and lexical boundary error type. Results indicate large inter-individual variability, but overall an increase in word identification accuracy and a decrease in lexical boundary errors in the auditory-visual condition. Further, some listeners made a greater proportion of lexical boundary insertions before strong syllables, suggesting that the addition of visual cues increased their use of syllable strength to identify word boundaries. Implications for clinical populations will be discussed.

**2pSCb24. Talker familiarity effects on speech-in-speech perception.**
Angela Cooper and Ann R. Bradlow (Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, akcooper@u.northwestern.edu)

Talker familiarity can facilitate the extraction of linguistic content from speech signals embedded in broadband noise; however, relatively little research has investigated the impact of talker familiarity with competing speech in the background. This study explores the effects of familiarity with the target or competing talker in speech-in-speech perception. Listeners were first familiarized with and trained to identify three female voices. They then completed a sentence recognition task in the presence of 1-talker babble. Familiarity with either the target or background talker was manipulated in separate conditions. Results revealed significantly better sentence recognition for familiar relative to unfamiliar target talkers in the presence of an unfamiliar background talker; however, sentence recognition with an unfamiliar target talker did not differ depending on background talker familiarity. Thus, while listeners were able to capitalize on familiarity with a talker's voice to aid target speech recognition, familiarity with the competing talker was neither facilitative nor inhibitory. This suggests that the influence of talker familiarity is limited to the actively attended stream. This stands in contrast to other aspects of the unattended stream which have been shown to exert an influence on speech-in-speech recognition including language and semantic content of the background speech.

**2pSCb25. Investigating the perception of relative speaker size using synthetic talkers.**
Santiago Barreda (Physiol., Univ. of Arizona, Dept. of Linguist., Edmonton, AB T6G 2E7, Canada, sbarreda@ualberta.ca) and Terrance M. Nearey (Linguist., Univ. of AB, Edmonton, AB, Canada)

Several previous experiments have investigated the perception of speaker size by presenting listeners with acoustic stimuli that differ in average f0 and/or apparent vocal tract length (i.e., higher formant frequencies overall), and asking listeners to make judgments of relative or absolute speaker size. Typically, these experiments use stimuli with a fixed phonetic content so that the acoustic characteristics of the stimuli may be compared directly. In this experiment, listeners were presented with pairs of vowels produced by synthetic speakers with different apparent vocal tract lengths and the same f0, and were asked to make judgments of relative size. However, listeners were presented with either the same, or different vowels produced by the two speakers. In some cases, differences associated with varying vocal tract lengths were in conflict with differences arising from the formant patterns associated with the differing vowel categories (e.g., lower F1 and F2 for /u/ vs /e/). Results suggest that judgments of relative size are affected by both the vowel categories presented and the apparent vocal tract lengths of the synthetic speakers. That is, more than a simple (category-corrected) vocal tract length estimate is involved in making size judgments for an unknown speaker.

**2pSCb26. The effects of listener biases on speech intelligibility.**
Jamie Russell (Linguist., Univ. of Br. Columbia, 9320 Dixon Ave., Richmond, BC V6Y1E7, Canada, jamie.russell@alumni.ubc.ca) and Molly Babel (Linguist., Univ. of Br. Columbia, Vancouver, BC, Canada)

Race, gender, and age have traditionally been considered the three big social categories perceivers attend to. Recent research has highlighted how

perceivers, listeners in this case, use the acoustic-phonetic information from spoken language to evaluate and categorize individuals. For example, studies have shown that different dialects elicit varying judgments of intelligence, friendliness and attractiveness. Such biases have implications regarding our assessment of individuals, be it in the clinic or the courtroom. While we know that our perception of speech can trigger biases, it is important to explore the opposite—whether our biases affect our perception of speech. This study aims to contribute to the knowledge base of the latter by examining the effects of listener biases on speech intelligibility. Listeners will be presented with sentences, independently controlled for clarity, produced by model talkers of different dialects and backgrounds. This auditory stimuli will be accompanied by an image or description of the talker as a method of eliciting or constructing a preconceived linguistic bias. After each sentence, participants will be asked to type out what they heard. Analyses will examine the proportion of correct words, confidence ratings, response time, and other measures exploring participants' ratings of vocal qualities.

**2pSCb27. Speaker normalization in noisy environments using subglottal resonances.** Harish Arsikere and Abeer Alwan (Elec. Eng. Dept., Univ. of California, Los Angeles, 56-125B Eng. IV Bldg., 420 Westwood Plaza, Los Angeles, CA 90095-1594, hari.arsikere@gmail.com)

This work investigates the use of subglottal resonances (SGRs) for speaker normalization in noisy environments. Based on our previous work, a noise-robust algorithm is developed for estimating the first three SGRs from speech signals; it achieves robustness by factoring the short-term (or local) signal-to-noise ratio into the estimation process. The SGR estimates provided by this algorithm are refined by applying maximum-likelihood (ML) corrections, and are used in a non-linear frequency-warping technique that we recently developed. This SGR-based normalization (SN) scheme is evaluated on the AURORA-4 database in clean and noisy conditions. Using power-normalized cepstral coefficients (PNCCs) as front-end features, SN reduces the average word error rate by 8.7% relative to ML-based vocal-tract length normalization (VTLN). A fast version of SN (without ML corrections of SGR estimates) is also found to outperform VTLN (by 5.9% relative); it is computationally less complex than VTLN and hence a potential alternative for real-time applications.

## Session 2pSP

## Signal Processing in Acoustics, Underwater Acoustics, Structural Acoustics and Vibration, and Animal Bioacoustics: Defense Applications of Acoustical Signal Processing

R. Lee Culver, Cochair
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Brian G. Ferguson, Cochair
*DSTO, PO Box 44, Pyrmont, NSW 2009, Australia*

**Chair's Introduction—1:25**

### *Invited Paper*

**1:30**

**2pSP1. Defense applications of acoustic signal time delay estimation.** Brian G. Ferguson and Kam W. Lo (Maritime Div., DSTO, PO Box 44, Pyrmont, NSW 2009, Australia, Brian.Ferguson@dsto.defence.gov.au)

Measurements of the differences in the times of arrival (or time delays) of a signal at spatially-separated sensors can be used to provide localization information about a source. For continuous broadband signals, the time delay for a given pair of sensors is estimated using the generalized cross-correlation method. The time delay estimate corresponds to the time lag at which the cross-correlation function attains its maximum value. The variance of the time delay estimates depends on the signal-to-noise ratio, signal bandwidth, and integration time. Errors in the source localization parameters depend on the variance of the time delay estimates as well as on the source-sensor geometry, stationarity of the sound propagation medium, uncertainty in the actual sensor positions, and the presence of multipath arrivals. Numerous practical examples of time delay estimation and the instantaneous localization of sources of military interest are presented for acoustic sensors deployed on land and under water. Also, localization performance is observed to degrade when the sound propagation medium becomes nonstationary, the direct path and multipath arrivals are unresolvable, or the differential Doppler effect is significant. Finally, the results of source motion parameter estimation using sequences of time delay estimates from pairs of sensors are presented.

### *Contributed Papers*

**1:50**

**2pSP2. Passive acoustic localization of small aircraft.** Alexander Sedunov, Alexander Sutin, Hady Salloum (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asedunov@stevens.edu), and Nikolay Sedunov (Stevens Inst. of Technol., Jersey City, NJ)

Stevens Institute of Technology has built the Acoustic Aircraft Detection (AAD) system for the detection, tracking and classification of Low Flying Aircraft (LFA). LFA may be of concern as they have been used for illicit operations. The AAD consists of several nodes deployed in a wide area, where each node acquires signals from a cluster of five microphones. The calculation of the cross-correlation function of acoustic signals received by various microphone pairs is applied for finding the direction of the signal arrival. Fusion of time difference of arrival estimates from several pairs of acoustic sensors has resulted in the improvement of angle estimation accuracy. Triangulation of the direction of arrival estimates from two or more nodes was applied for determining the target position and altitude. Kalman filtering was used for smoothing the target tracks and decreasing the localization uncertainties. Software for predicting the performance was developed and was used to inform sensor placement in the field tests. The field tests were conducted with various kinds of LFA—single-engine, helicopters, and ultralights. Comparison of the acoustic tracking results with the GPS ground truth showed that the errors are similar to the theoretical predictions. [This work was funded by DHS S&T.]

**2:05**

**2pSP3. Automated acoustic detection and classification of small aircraft.** Yegor Sinelnikov, Alexander Sutin, Alexander Sedunov, and Hady Salloum (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asedunov@stevens.edu)

Automated acoustic detection and classification of low flying aircraft can be used for the prevention of illicit operations. In this presentation, we review past techniques and a new acoustic classification and detection technique. The sound detection algorithm is based on locking in narrow frequency components in the spectra of the recorded aircraft sound. The classification algorithm uses tonal components in the spectral and cepstral domains using discrete time window peak picking. A set of chosen classifiers include: fundamental (pitch) frequency extracted by the cepstral analysis, frequency of tonal components with the maximal amplitude, power spectral estimate in various frequency bands, and number of peaks in predefined frequency and frequency windows. Various small single engine aircraft, ultralights, and helicopters are acoustically detected and classified as they approach the sensor. The algorithm enables simultaneous target detection, reduces noise sensitivity, and minimizes classifier feature space, while maintaining good classification separation. The acoustic classification algorithm was incorporated into the Acoustic Aircraft Detection system developed by Stevens. This combination allowed automated Doppler correction of the harmonic lines, the optimal part of the aircraft signal selection, and establishing classification merits based on aircraft track. [This work was funded by DHS S&T.]

2:20

**2pSP4. Coordinated optimization of stationary and moving sensors with scenario-specific coverage and cost goals.** Sergey Vecherin, D. Keith Wilson (Signature Phys. Branch, U. S. Army ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, Sergey.N.Vecherin@usace.army.mil), and Chris Pettit (Aerosp. Eng. Dept., U. S. Naval Acad., Annapolis, MD)

The problem of simultaneously optimizing coverage among stationary sensors placed on the ground and moving sensors on aircraft is considered. The formulation extends a previously developed algorithm for optimizing placement of ground sensors subject to signal propagation effects and terrain-dependent coverage preferences. To this end, candidate aircraft routes should be characterized in terms of cost and coverage. Cost can reflect a variety of disincentives, for example, the probability that an aircraft can be heard on the ground. Coverage reflects the area of regions with probability of detection higher than a specified threshold. The extended algorithm can be applied to a variety of planning scenarios, such as determining an optimal combination of routes for multi-aircraft operations, optimization of aircraft routes to supplement ground-sensor coverage, optimization of ground sensors to cover blind spots of aircraft coverage, and simultaneous optimization of ground sensors and aircraft coverage. An illustrative problem of routing an unmanned aircraft system (UAS) to provide surveillance around a roadway while minimizing aircraft audibility at specified locations on the ground is considered in detail.

2:35

**2pSP5. Performance comparison of distributed and centralized systems based on information.** Tsih C. Yang (College of Marine Sci., Nat. Sun Yet-sen Univ., 70 Lien Hai Rd., Kaohsiung 804, Taiwan, tsihyang@gmail.com)

Distributed systems using many sensors widely distributed over a large area present an alternative way for target detection and environmental sensing. The diversity of opportunities for detection by widely distributed sensors seems attractive but the signals received on distributed sensors are usually not coherent due to the wide separations between the receivers. Hence, conventional approaches based on signal coherence/gain no longer apply. In this paper, information theory will be used to compare their performance. Treating the target radiated signal as a communication signal, transmitting continuous Gaussian-distributed alphabets, the Shannon channel capacity yields the maximum information that the receivers can learn about the (target) transmitted signal. The channel capacity can then be used as a metric to compare the performance of various sensor systems in the ideal case. Matched tracking processing is introduced to motivate a capacity-based detector and detection capacity. Based on that, it is found that the distributed systems can achieve in principle an area of coverage two to three times larger than that of a centralized system under the right conditions, and the area of coverage by the entire system can be significantly larger than the sum of detection areas of individual nodes.

**2:50–3:05 Break**

3:05

**2pSP6. Acoustic signatures of small boats measured in controlled lake conditions.** Alexander Sutin, Michael DeLorme, Hady Salloum, Alexander Sedunov, Nikolay Sedunov, Robert Weiss, Mikhail Tsionskiy, and Howard Goheen (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Measurements were conducted under controlled conditions at Lake Hopatcong, NJ, to compare acoustic signatures of various small boats. The tested boats included: an outboard-driven Panga, a multiple-outboard driven go-fast boat, an electric drive vessel, a personal watercraft (PWC), and two small outboard-driven rigid hull inflatable boats (RIBs). Stevens Passive Acoustic Detection System (SPADES) was used for acoustic measurements and the specially developed "Portable Vessel Data System" (PVDS) was used to record vessel position, speed, shaft RPM, and vessel orientation. Vessel acoustic Source Level measurements were conducted by comparing the recorded vessel acoustic signals with a signal generated by a calibrated emitter from the same point. Dependencies of acoustic signatures on boat speed and loading were also investigated. For several vessels, the Source Level decreased when its speed increased. Analysis of tonal components in the vessels acoustic signatures with Detection of Envelope Modulation on Noise (DEMON) allowed to determine the number of engines per boat and even the gear ratio for transformation of shaft rotation to the propeller rotation. [This work was supported by DHS S&T.]

3:20

**2pSP7. Active–passive acoustic system for underwater port protection.** Hady Salloum (Stevens Inst. of Technol., Hoboken, NJ), Andrew Meeecham (Sonardyne, Inc., Newport, RI), and Alexander Sutin (Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, asutin@stevens.edu)

Active and passive acoustic systems may be used for port protection against threats from divers, free swimmers, Unmanned Underwater Vehicles, etc. Currently, underwater port protection is provided by Diver Detection Sonars (DDS), and Sonardyne has developed the market-leading active system, Sentinel. Stevens' scientific research in passive acoustics led to the development of algorithms for threat detection that were realized in the Stevens Passive Acoustic Detection System (SPADES). Sonardyne and Stevens have formed a partnership to investigate a combined active/passive system that will leverage the functionality of existing sensors to design, develop, and produce a system with superior capabilities and performance as opposed to a single system. The combined system provides detection not only for underwater targets, but also for surface vessels. Possible configurations include: (1) a scenario where acoustic pulses emitted by the Sonardyne sonar and reflected by the target can be received and processed by SPADES; (2) an "acoustic fence," whereby the detection of targets occurs as they cross a line between the transmitter and receiver. This configuration takes advantage of the significant increase in target scattering strength in the forward direction versus that from the backscatter; and (3) a fusion between detections and/or tracks from the passive and active sensors to provide an increased probability of detection and reduced probability of false alarm.



3:35

**2pSP8. Practical applications of track segment association algorithms to an active sonar network for underwater port surveillance.** Patrick Edson and Peter J. Stein (Scientific Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, pedson@scisol.com)

Acoustic detection and tracking of small (swimmer-size) targets while minimizing the incidents of false alerts can be challenging in a shallow, multipath, high-clutter harbor environment. One common problem involves track intermittency where periods or locations of high clutter or multipath interference inhibit detection and cause tracks to fail and restart. This paper describes the results of applying track segment association and related algorithms from airborne and ground radar applications [*e.g.,* Zhang and Bar-Shalom, IEEE Trans. Aerosp. Electron. Syst. **47**(3), 1899–1914 (2011)] to an active sonar network for underwater port surveillance [Edelson *et al*., J. Acoust. Soc. Am. **129**(4), 2598 (2011), Stein *et al*., J. Acoust. Soc. Am. **121**(5), 3084 (2007)] using real-world swimmer and synthetic target data.

3:50

**2pSP9. Adaptive processing for mitigation of biologically induced active-sonar clutter and reverberation.** Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Scattering from biological entities can be a significant source of reverberation and clutter for active sonar systems operating in deep or shallow water. Resonant scattering from the swim bladders of fish is a dominant source of this biologically induced reverberation and clutter for midfrequency active sonar (1–10 kHz), which can obscure targets, result in false targets, and otherwise overload operators of tactical systems. Though significant questions remain regarding the fisheries oceanography needed to predict clutter and reverberation, the basic physics of scattering from individual fish and groups of fish is well understood. This motivates a data-driven, clutter adaptive approach to mitigation of biologically induced reverberation and clutter that utilizes physical models without requiring unavailable databases of fish-species abundance and local spatiotemporal distributions. In this work, physics-based models relating physical parameters of fish and their spatiotemporal behaviors to spectral properties and spatiotemporal distributions of scattering form the basis of *in situ* parameter estimation. In this presentation, adaptive processing developed for the early stages of the active-sonar processing chain using these *in situ* estimates is described, including waveform and matched-filter design and within and across-beam normalization. [Work supported by a NAVSEA Phase I SBIR award.]

**4:05**

**2pSP10. Accuracy and resolution of bearing measurements for crossed-dipole receiver.** Chunsheng Liu and Mark V. Trevorrow (Defence R&D Canada Atlantic, 9 Grove St., Dartmouth, NS B2Y 3Z7, Canada, chunsheng.liu@drdc-rddc.gc.ca)

A well-known system for estimating the bearing to an underwater acoustic source consists of two crossed dipole sensors and an omni-directional sensor. At DRDC Atlantic, several bearing estimation methods for this three-channel system in the presence of ocean noise have been examined with the goal of minimizing the bearing error. Example algorithms are arctangent and maximum likelihood beamformer. These algorithms have trade-offs between estimation accuracy and computational efficiency, because any practical bearing-estimation algorithm must remain efficient enough for implementation in a real-time sonar processor. The accuracy of these algorithms can be demonstrated through comparison with the Cramér-Rao lower bound. Previously the calculation of bearing accuracy was based on the assumption of isotropic noise and a single signal source. However, the case of anisotropic noise and multiple-sources is more realistic, especially for bistatic or multistatic active sonar. Adaptive methods such as the Minimum Variance Distortionless Response (MVDR) beamformer will be explored to address this problem. The paper will discuss the potential improvement of bearing estimate accuracy and bearing resolution through numerical models.

**4:20**

**2pSP11. Fast nearfield to farfield conversion for circular synthetic aperture sonar.** Daniel Plotnick and Phillip L. Marston (Washington State Univ., 1510 NW Turner Dr., Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com)

Monostatic circular synthetic aperture sonar (CSAS) images are formed by processing azimuthal angle dependent backscattering from a target at a fixed distance from a collocated source/receiver. In the laboratory data is taken by fixing the source location and spinning the target via a rotating mount. Typical CSAS imaging algorithms [Marston *et al*., Proc. IEEE Oceans (2011); Ferguson *et al*., J. Acoust. Soc. Am. **117**, 2915 (2005)] assume the scattering data are taken in the farfield. Experimental constraints may make farfield measurements impractical and thus require a target to be scanned in the nearfield. If left uncorrected, this results in distortions of the target image and possible distortions of the angular dependence of features. A fast approximate Hankel function based algorithm is presented that converts nearfield data to farfield data. Images and spectrograms of an extended target are compared for both cases. Spatial sampling requirements for data collection are also considered. [Work supported by ONR.]

TUESDAY AFTERNOON, 3 DECEMBER 2013 CONTINENTAL 6, 2:00 P.M. TO 4:25 P.M.

## Session 2pUW

## Underwater Acoustics: Sound Propagation Through and Scattering by Internal Waves, Spice, and Finestructure in Shallow Water II: Past, Present, and Future

Steven I. Finette, Cochair
*Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320*

James Lynch, Cochair
*Woods Hole Oceanogr., M.S. # 11, Bigelow 203, Woods Hole, MA 02543*

### *Invited Papers*

**2:00**

**2pUW1. Predicting internal waves of acoustical significance.** Timothy F. Duda (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, tduda@whoi.edu)

The broad dynamic range of internal gravity wave periods and scale lengths, as well as the variable group geometries of these waves, means that they are responsible for a variety of acoustic propagation effects. As a result, their degree of acoustical significance and the specific effects that they cause depend on time and place. A variety of internal-wave (IW) acoustic effects will be reviewed, with analysis of the regimes where each effect may have first-order propagation consequences. Many of the effects are specific to details such as the IW amplitudes, the angle between acoustic and IW propagation paths (and thus IW direction), IW proximity to sound source and/or receiver, and whether the IW field can be considered to be a random medium or, alternatively, a set of identifiable scattering features. The current state of our knowledge motivates a question: To what degree are specific acoustically relevant aspects of the IW field, and thus the acoustic effects, predictable? The answer to this question is being investigated through efforts to build ocean models bridging large-scale ocean dynamics, smaller-scale IW, and submesoscale dynamics, plus two-dimensional and three-dimensional acoustic propagation behavior.

2:20

**2pUW2. Anisotropy in horizontal plane for the sound propagation in the ocean in the presence of internal waves.** Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru)

Review of researches dedicated to the sound propagation in the presence of internal waves is presented (starting with Evgeny Malyuzhinets, 1959 and Owen Lee, 1961). Main attention is paid on dependence of properties of the sound field (for example, sound amplitude fluctuations) on direction of propagation in horizontal plane. It was established—three main mechanisms: mode coupling, adiabatic variations, and horizontal refraction can be manifested in dependence on direction of sound propagation relative wave front of internal waves. Theoretical results as well as results of numerical modeling are presented illustrating different properties of acoustical signals for mentioned mechanisms including dependence on mode number and frequency. Experimental observations are also presented confirming real manifestations of mentioned three mechanisms.

## *Contributed Papers*

2:40

**2pUW3. Modeling the nonlinear internal wave field for coastal acoustics.** Arthur E. Newhall, Ying-Tsong Lin, James F. Lynch, Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., 210 Bigelow Lab., M.S. #11, Woods Hole, MA 02543, anewhall@whoi.edu), and Karl R. Helfrich (Physical Oceanogr. Dept., Woods Hole Oceanographic Inst., Woods, MA)

It is now well known that coastal nonlinear internal waves, a common ocean feature, have large effects on the propagation of sound in the coastal ocean. Modeling such waves for input to acoustic models using regional ocean numerical models is difficult due to the fine spatial and temporal resolution needed. A multi-university program called Integrated Ocean Dynamics and Acoustics (IODA) is addressing this problem using two approaches—one a two-way, direct nested ocean model approach, and the other a blending of mesoscale ocean models and fine scale internal wave models. In this talk, we will discuss the latter approach. Issues of the internal wave source positions, their nonlinear propagation, earth rotation effects, field interpolation, output verification and integration of the ocean model with acoustics codes will be discussed. The results going from an ocean model to fully 3-D acoustic fields will be presented. [Work sponsored by ONR MURI program.]

2:55

**2pUW4. Effect of internal waves on the waveguide invariant distribution.** Daniel Rouseff (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St,, Seattle, WA 98105, rouseff@apl.washington.edu)

Multipath propagation in the ocean waveguide results in constructive and destructive interference between the paths. The resulting interference pattern, mapped versus range and frequency, often exhibits striations, alternating bands of high and low intensity. In simple waveguides, the trajectory of these striations can often be described by a single scalar parameter, the so-called waveguide invariant. In more complicated waveguides, the concept can be generalized to create the waveguide invariant distribution. In the present work, the effects of shallow water internal waves on the waveguide invariant distribution are examined. It is shown how internal waves can blur otherwise sharp striations and flatten the waveguide invariant distribution. [Work supported by ONR.]

3:10

**2pUW5. Modal interference and frequency striations induced by moving nonlinear internal waves.** Tsih C. Yang (College of Marine Sci., Nat. Sun Yet-sen Univ., 70 Lien Hai Rd., Kaohsiung 804, Taiwan, tsihyang@gmail.com) and Jin-Yuan Liu (Nat. Taitung Univ., Taitung, Taiwan)

Spectrogram of a broadband signal propagated over distances often displays striated bands of constant acoustic intensity levels plotted against the range and frequency. This phenomenon is due to modal interference and the slope of the striation is related to the source range and frequency by the waveguide invariant parameter “beta.” As an application, one can estimate the range to a moving source based on the striation of the spectrogram (as a function of frequency and time), knowing the source velocity and the waveguide invariant parameter. Conversely, no frequency striation is expected in the spectrogram of the received signal when both the source and receiver are fixed (the frequency content remains unchanged with respect to time). However, when nonlinear internal waves, appearing as a train of moving solitons, are present in the propagation path, the spectrogram of the received signal will display frequency striations (with respect to time) even for a fixed source and receiver. This paper applies the mode coupling equation to calculate the modal interference, intensity fluctuation and the striation slope. Broadband spectrograms are simulated to show the frequency striation as the solitons move from the receiver to the source. Results from at-sea data will be presented for comparison.

3:25

**2pUW6. Synthetic aperture sonar imaging of breaking internal wave structures.** Anthony P. Lyons (Appl. Res. Lab, Penn State Univ., University Park, State College, PA 16803, apl2@psu.edu), Roy E. Hansen, Torstein O. Sæbø (Norwegian Defence Res. Establishment (FFI), Kjeller, Norway), and Hayden J. Callow (Kongsberg Maritime, Horten, Norway)

In October 2012, the Centre for Maritime Research and Experimentation (CMRE) conducted the ARISE’12 trials from the NATO research vessel Alliance, off of Elba island, Italy. During this trial, data were collected by the Norwegian Research Defence Establishment (FFI) using a HUGIN AUV with interferometric synthetic aperture sonar (SAS). Large visible structures in the SAS images and in the SAS bathymetries were caused by features in the water column, called boluses, which formed after breaking internal wave events. Variation in backscattered intensity from the seabed was caused by focusing of sound by the bolus and errors in bathymetry by the changes in phase as the acoustic energy moved through the lower sound speed bolus. A ray model and a 3D parabolic equation (PE) model were used to simulate the effects of the bolus on the acoustic field incident on the bottom and results compared favorably with SAS data. Models for the change in bolus size and speed as it propagated upslope also gave reasonable comparison to estimates obtained from SAS intensity images. [The work performed by APL was supported by the US Office of Naval Research.]

3:40

**2pUW7. Modeling sound propagation through internal waves using a spectral element method.** Sumedh M. Joshi (Ctr. for Appl. Mathematics, Cornell Univ., 657 Rhodes Hall, Ithaca, NY 14850, sumedh.m.joshi@gmail.com), Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Peter J. Diamessis (Dept. of Civil and Environ. Eng., Cornell Univ., Ithaca, NY)

Considered here is the problem of low-frequency sound propagation over shallow, shoaling bathymetry in the presence of perturbations to the background sound velocity profile due to internal waves (IW). The question we attempt to answer is: to what degree can heuristic models of IWs coupled to numerical sound propagation models capture the variability in sound propagation observed in the environment? A high-order finite element model is employed to compute the acoustic field as it propagates through these IWs. The generality of the finite element method allows for spatial and temporal sound speed variations, and its convergence properties yield arbitrarily small error as the grid is rened. Simulations in the presence

and absence of IWs will demonstrate the degree to which IWs influence sound propagation. Different models of IWs will demonstrate the sensitivity of the sound propagation to the choice of heuristic used for the IWs. Results will be shown for shoaling waveguides of O(100 m) depth and O(10 km) range, and frequencies of O(50 Hz). [Work supported by the NDSEG Fellowship.]

**3:55**

**2pUW8. From Swarm 1995 to Shallow Water Experiment 2006: What have we learned in shallow waveguide acoustics?** Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Last 20 years have witnessed some very interesting scientific discoveries in shallow water waveguide physics. While before1995 SWARM experiment most studies in shallow water waveguides considered two-dimensional (2D) slices of the ocean waveguide to describe the behavior of sound wave propagation, many studies in the recent years have focused on the three dimensionality of the acoustic wave propagation in the presence of water column variability. This paper summarizes the highlights of what has been done with reference to the published literature and some ongoing work. [Work supported by ONR322OA.]

**4:10**

**2pUW9. Estimates of source range using horizontal multipath in continental shelf environments.** Megan S. Ballard (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

In a continental shelf environment, energy from an acoustic source propagating obliquely upslope repeatedly reflects from the sea surface and sloping seafloor with a consequent change in direction with each bottom reflection. Measurements of this type of effect can be observed in beamformed data from a horizontal line array. The arrivals from a single source are seen on two beams: a direct path with bearing angle corresponding to the source location and a refracted path with a bearing angle inshore of the source. In this work, the horizontal multipath effects are exploited to estimate the location of an acoustic source. Using the hybrid modeling approach of vertical modes and horizontal rays, rays are traced in the horizontal plane with refraction determined by the modal phase speed. Invoking reciprocity, the rays orientate from the center of the array and have launch angles equal to the estimated bearing angles of the direct and refracted paths. The location of the source in the horizontal plane is estimated from the point where the rays intersect. The technique is applied to data recorded on a horizontal line array located about 12 km east of the southern coast of Florida. [Work supported by ONR.]

TUESDAY AFTERNOON, 3 DECEMBER 2013 UNION SQUARE 14, 1:45 P.M. TO 3:15 P.M.

## Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R.J. Peppin, Chair ASC S1
*5012 Macon Road, Rockville MD 20852*

A. Scharine, Vice Chair ASC S1
*U.S. Army Research Laboratory, Human Research & Engineering Directorate*
*ATTN: RDRL-HRG, Building 459 Mulberry Point Road*
*Aberdeen Proving Ground MD 21005 5425*

**Accredited Standards Committee S1 on Acoustics**. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound, etc. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

*People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.*

**Scope of S1**: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance and comfort.

## Meeting of Accredited Standards Committee (ASC) S12 Noise

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

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

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

*People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note - that meeting will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.*

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



TUESDAY EVENING, 3 DECEMBER 2013 8:00 P.M. TO 10:00 P.M.

## OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Tuesday are as follows:

| | |
|---|---|
| Acoustical Oceanography | Plaza B |
| Animal Bioacoustics | Union Square 23/24 |
| Architectural Acoustics | Golden Gate 4/5 |
| Engineering Acoustics | Mason |
| Noise | Continental 9 |
| Physical Acoustics | Continental 7/8 |
| Psychological and Physiological Acoustics | Continental 2/3 |

# Session 3aAAa

## Architectural Acoustics: Separating Spaces: Adventures in Acoustic Isolation of Acoustically Sensitive Spaces

Shane J. Kanter, Cochair
*Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604*

Stephanie Hoeman, Cochair
*Environmental Systems Design Inc., 175 W. Jackson Blvd., Chicago, IL 60604*

**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**3aAAa1. An examination of the minimum construction necessary to improve the sound isolation in a university's existing music building.** Ashley Masoner (Acoust. Dimensions, 120 N Pelham Rd., Apt. 1L, New Rochelle, NY 10805, amasoner@gmail.com)

The National Association of Schools of Music (NASM) is the organization responsible for accrediting both degree- and non-degree granting music schools in the United States. Typically, NASM accreditation is based on the musical program itself, but what happens when an existing building's sound isolation is so poor it threatens a school's accreditation status? As is often the case in the Arts, the University of Wyoming has a limited budget to use for upgrades in their existing Music building. Through a series of mock-up tests, Acoustic Dimensions was able to identify the most critical noise-flanking paths and recommend the most cost effective means of raising the NIC values between practice rooms to levels that better accommodate simultaneous use of adjacent spaces.

**8:25**

**3aAAa2. Night club skylight vs. hotel—A case study.** Shane J. Kanter, Carl Giegold, Constance Walker, and John Strong (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, skanter@thresholdacoustics.com)

The Paris Club, a popular night club in Chicago, Illinois, is enclosed by an enormous, operable skylight. This feature spans across the entire space, acting as the only "barrier" between the club and the directly adjacent hotel and high end condominium buildings. From within the Paris Club, the condominium balconies and hotel windows are in direct line-of-site. Although the skylight is a wonderful architectural feature, it is devastating to the isolation of thumping music played within the club. Through various testing methods, the isolation performance was measured and required performance was determined. Calculations and a laboratory tested mockup yielded a viable solution.

**8:45**

**3aAAa3. The isolation of Lookingglass Theatre, or how an active municipal pumping station became the site for a successful theater.** Gregory A. Miller (Threshold Acoust., LLC, 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, gmiller@thresholdacoustics.com and Richard H. Talaske (TALASKE/Sound Thinking, Oak Park, IL)

In search of a new permanent home, the Lookingglass Theatre Company was offered the chance to transform a former steam generation plant within the Chicago Avenue Pumping Station into a multi-venue theater. The Pumping Station is a historic structure—one of the few to survive the 1871 Chicago Fire—but remains an active part of the municipal water supply, pumping water to the near north side of the city. The pump room is less than 5 ft. from the site of the main theater, a 270-seat reconfigurable space specified to achieve a background noise level of RC-25. This paper will describe design challenges identified for achieving this level of acoustic separation, the construction challenges encountered building within a 130-year-old historically protected structure, and the successful result achieved by overcoming these challenges.

**9:05**

**3aAAa4. Design and implementation of an open-plan art gallery with heavily amplified audio/video works.** John T. Strong and Carl P. Giegold (Threshold Acoust., 53 W. Jackson Blvd., Ste. 815, Chicago, IL 60604, jstrong@thresholdacoustics.com)

An exhibit consisting of short film and video works by internationally recognized artist and film director Steve McQueen was installed in a large open-plan gallery at the Art Institute of Chicago. Several of the works included a heavily amplified audio component with very high levels of low-frequency energy, such as machine noise and hip-hop music. The layout of the gallery dictated that these works be placed in close proximity to each other as well as to works requiring a very low noise floor, while maintaining an open-plan

format within an existing gallery space. Temporary enclosures were designed and constructed to allow these works to be displayed concurrently without significant disturbance to each other. Providing favorable room acoustics for the heavily amplified program within each space was also of primary importance. Panel absorbers tuned to the modal characteristics of each space as well as the unique acoustic characteristics of each work were integrated into the construction along with diffusive and absorptive surfaces to provide a comfortable listening environment for listeners while maintaining a uniform appearance.

**9:25**

**3aAAa5. Two case histories relating to the transmission of noise from mechanical equipment rooms to adjacent noise sensitive spaces.** Robert C. Coffeen (Architecture, Univ. of Kansas, Marvin Hall, 1465 Jayhawk Blvd., Lawrence, KS 66045, coffeen@ku.edu)

Two recent noise control projects related to the reduction of noise from mechanical equipment rooms to an adjacent conference room and to an adjacent office. For both of these situations, the noise control procedures to be implemented involved the determination of the most significant noise transmission path; air borne noise transmitted through a concrete floor slab or structurally transmitted noise due to lack of proper chiller and pipe vibration isolation in the conference room case, or for the office case airborne noise transmitted through a wall constructed of gypsum wall board and flanking through common floor slabs or structurally transmitted due to lack of suitable pump and piping vibration isolation. A method for determining the most significant noise transmission path and the resulting noise reduction for these two cases will be described.

**9:45**

**3aAAa6. Case study: Isolation of a television production studio from airborne noise and ground-borne vibration.** Eric L. Reuter (Reuter Associates, LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

This project involved design of a 650-s.f. television and audio production studio in an existing historic building situated along a busy urban street with a subway tunnel below. Low background noise in the space was the primary concern, but the possibility of noise from music performance in the space interfering with nearby editing rooms and on-air studios was also considered. In order to satisfy all of the isolation requirements, the studio was constructed as a free-standing structure, supported by spring isolators on a new steel structure, within the existing building shell.



## *Contributed Papers*

**10:05**

**3aAAa7. Upgrading secret military facilities—What is more important, acoustic design standards or acoustical performance?** Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

The U.S. Department of Defense has developed acoustical performance standards that are to be achieved in the design and construction of meeting and conference rooms where sensitive and secret information will be discussed. These performance standards rely on published acoustical industry design data, which are readily available. The intention of these standards is to prevent sensitive and secret information from being heard, understood, or otherwise obtained by persons or devices that are not authorized to have access to such information. This paper presents design and field performance test results for new and renovated secret rooms that initially passed the acoustical design criteria and acoustical standard field tests, but failed to provide the desired secret level acoustical performance. Further investigations and research into partition, component, and building composite performance indicated that floors, walls, ceilings, doors, windows, and perimeter penetrations by conduit and HVAC ducting, which individually met the design standards and when installed meet the design standards, but as a composite did not provide the intended acoustical performance that would prevent unauthorized access to sensitive and secret information by persons or devices outside the designated perimeter. Reasons for certain performance failures are discussed and specific successful remedies are presented.

**10:20**

**3aAAa8. Development of a low-frequency impact noise metric.** David W. Dong and John LoVerde (Veneklasen Associates, 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com)

Low frequency footfall noise (“thudding”) is a common source of complaints in lightweight joist-framed multifamily projects. Previous work by the authors has indicated that the low frequency impact sound pressure levels (LFISPL) from a standard ISO tapping machine are highly correlated with occupant reaction [LoVerde and Dong, J. Acoust. Soc. Am. **112**, 2201 (2002), LoVerde and Dong, INCE Inter-Noise 2004 Proceedings, **167** (2004)]. In order to be useful, the raw LFISPL must be translated into a single number that maintains the high correlation with subjective reaction while providing adequate dynamic range to distinguish the performance of different assemblies. Candidate metrics are evaluated.

**10:35**

**3aAAa9. Acoustical characteristics of restorative space on a university campus.** Abigail Bristow (School of Civil and Bldg. Eng., Loughborough Univ., Loughborough LE11 3TU, United Kingdom, a.l.bristow@lboro.ac.uk) and Kirill V. Horoshenkov (School of Eng., Univ. of Bradford, Bradford, United Kingdom)

This paper explores the nature, sound environment, and value of restorative space on University campuses in the United Kingdom that is separated from standard teaching venues. Questionnaire surveys were undertaken in the Atrium and Student Central at the University of Bradford and in the Library at Loughborough University to explore expectations and perceptions of restorative quality, preferences, values, and use on non-teaching space. These spaces are used routinely by the students for social, relaxation, individual, and group studies. Contemporaneous, continuous measures of noise were undertaken at the survey locations. Noise levels in a social space on campus can be as high as 85 dB, which affects the restorative quality of the space. The results of these surveys enabled us to explore the link between the acoustical characteristics of a space and its restorative quality, and to identify key sounds which contribute to the value of a restorative space. The value of such spaces is reflected in the willingness to pay for quieter and greener spaces.

# Session 3aAAb

## Architectural Acoustics and Noise: Restaurant Acoustics

Eric L. Reuter, Cochair
*Reuter Associates, LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801*

Steven D. Pettyjohn, Cochair
*The Acoustics & Vibration Group, Inc., 5700 Broadway, Sacramento, CA 95820*

**Chair's Introduction—10:55**

### *Invited Papers*

**11:00**

**3aAAb1. Design of a new restaurant and remedial design of an existing café: Reducing noise of patron activities.** Steve Pettyjohn (The Acoustics & Vibration Group, Inc., 5765 9th Ave., Sacramento, CA, spettyjohn@acousticsandvibration.com)

R. Selands is an experienced restauranteur with an understanding of the importance of controlling excess sound, as was his architect from the Netherlands. When designing a new restaurant in Sacramento called Ella, acoustics was an important item. Both sound from the HVAC system and patron noise were evaluated. Acoustical treatment of patron noise had to fit in with the aesthetic plan by the architect. HVAC sound was controlled using specific wall and floor/ceiling design and spring isolation. The result was very successful with continuous compliments from patrons. Some issues still exist where a decision was made to not use acoustical treatment. When designing a new market and café, acoustics was not high on the list until the day the facility was opened. Within three days, calls were made requesting assistance to reduce or eliminate excess sound because of complaints from patrons. Site visits, limited field tests, and drawings were used to evaluate the architectural acoustics. Options were provided to modify room finishes to reduce the reverberation time within the space. The most difficult task was modifying the tin-type ceiling to allow sound absorptive material to be placed above.

**11:20**

**3aAAb2. Disco dining: Where DJ culture meets high-end restaurants.** Tyler Adams (JBA Consulting Engineers, 36 Technol. Dr. Ste. 200, Irvine, CA 92618, tadams@jbace.com) and Michael Schwob (JBA Consulting Engineers, Las Vegas, Nevada)

Live DJs have quickly become a popular feature in the contemporary landscape of high-end restaurants. Many new establishments are sacrificing prime dining floor area so they may prominently feature a DJ booth. The Restaurateur's goal is not simply to entertain patrons and create ambiance but to use the name-recognition of DJs to draw customers and impart a sense of luxury and "cool" cachet. Along with this cultural phenomenon comes an assortment of sound isolation and room acoustics issues that must be addressed to provide patrons with an enjoyable dining and listening experience and ensure the thumping dance music results in minimal impact to adjacent spaces and properties. In this talk, a number of such issues will be discussed using real-world problems and solutions from current high-end restaurants.

**11:40**

**3aAAb3. Examining the relationship between room acoustics parameters and the cafe effect in restaurants.** Eric L. Reuter (Reuter Associates, LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com) and Christopher O'Connor (Berklee College of Music, Boston, MA)

It is well known that the so-called café effect, the progressive increase in speech effort as an enclosed space becomes more crowded with talkers, is related to the reverberant characteristics of the room. This study attempts to correlate specific room acoustics parameters (total absorption, reverberation time, etc.) to the café effect through analysis of several existing restaurants. At each establishment, noise monitoring was conducted over the course of a busy weekend, followed by corresponding impulse response measurements in the empty room. Results and conclusions from the study will be presented.

## Session 3aAB

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

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

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

### *Invited Papers*

**8:00**

**3aAB1. Hierarchical effects of attention on amplitude modulation encoding in auditory cortex.** Mitchell Sutter, Kevin N. O'Connor, Joshua Downer, Jeffrey Johnson, and Mamiko Niwa (Ctr. for Neurosci., Univ. of California Davis, 1544 Newton Cr, Davis, CA 95618, mlsutter@ucdavis.edu)

How attention influences single neuron responses in the auditory system remains unresolved. We found that when monkeys actively discriminated temporally amplitude modulated (AM) from unmodulated sounds, primary auditory (A1) and middle lateral belt (ML) cortical neurons better discriminated those sounds than when the monkeys were passively listening. This was true for both rate and temporal codes. Differences in AM responses and effects of attentional modulation on those responses suggest: (1) attention improves neurons' ability to temporally follow modulation (2) non-synchronized responses play an important role in AM discrimination (3) ML attention-related increases in activity are stronger and longer-lasting for more difficult stimuli consistent with stimulus specific attention, whereas the results in A1 are more consistent with multiplicative nonlinearity, and (4) A1 and ML code AM differently; ML uses both increases and decreases in firing rate to encode modulation, while A1 primarily uses activity increases. These findings provide a crucial step to understanding both how the auditory system encodes temporal modulation and how attention impacts this code. Further, our findings support a model where rate and temporal coding work in parallel, permitting a multiplexed code for temporal modulation. [Work supported by NIDCD RO1 DC-02514.]

**8:20**

**3aAB2. Forebrain processing of complex sounds: From mice to humans.** Christoph E. Schreiner, Craig Atencio, Jonathan Shih, Patrick Hullett (Otolaryngol., UCSF, 675 Nesln Rising Ln., San Francisco, CA 94143-0444, chris@phy.ucsf.edu), and Edward Chang (Neurological Surg., UCSF, San Francisco, CA)

The decomposition and re-integration of complex sounds, such as speech, are at the core of auditory cortical processing. We will discuss the transformation of this process between subcortical and cortical stations from single-filter to multiple-filter models and its relationship to the structural organization of auditory cortex in animal models. We will compare findings of recordings from human superior temporal gyrus to speech sounds with the findings obtained from animal models and discuss potential implications for speech representation in primary and non-primary cortical areas.

**8:40**

**3aAB3. Individual identification of Japanese macaques by coo-calls: Pitch or vocal tract characteristics?** Takafumi Furuyama, Kohta I. Kobayasi, and Hiroshi Riquimaroux (Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp)

Japanese macaques, *Macaca fuscata*, utter the Smooth Early High (SEH), one of harmonically structured coo-calls, for greeting and locating other individuals. The purpose of this study was to examine acoustical features of SEH for them to identify the caller. Two male Japanese macaques were trained to discriminate SEHs of Macaque A (SEHa) from those of B (SEHb). The fundamental frequencies (pitch) of SEHa and SEHb were different. Those calls were recorded from monkeys unfamiliar to subjects. GO-NOGO paradigm was used for training the animals. SEHa served GO stimuli (S+) while SEHb served NOGO stimuli (S-). A "Hit" was reinforced with about 1 ml of orange juice. Unfamiliar SEHa and SEHb, which were never used for training sessions, were used as test stimuli. A series of morphed SEHs between SEHa (S+) and SEHb (S-) with only pitch or vocal tract characteristics varied were also used for test stimuli. Results showed that reaction times (RTs) for unfamiliar SEHa were significantly shorter than those for unfamiliar SEHb. RTs to morphed stimuli shortened as the distance from SEHb (S-) increased in both pitch and vocal tract dimensions. Data suggested that both pitch and vocal-tract characteristics were important for Japanese macaques to identify individuality.

**9:00**

**3aAB4. Early experience improves neural discrimination/recognition of natural complex sounds.** Shaowen Bao (Helen Wills Neurosci. Inst., Univ. of California, 210X Barker Hall, Berkeley, CA 94720, sbao@berkeley.edu)

In natural environments, behaviorally relevant complex sounds are often produced with "speaker" variability and contaminated with environmental noises. To efficiently discriminate/recognize natural complex sounds, the auditory system has to tune to defining acoustic features and filter out random, meaningless features. In humans, for example, efficient speech recognition is achieved by enhancing perceptual contrast for native speech sounds, as well as reducing perceptual contrast for non-native speech sounds. The neural mechanisms underlying this perceptual transformation are still not well understood. We exposed juvenile rats to heterospecific vocalizations recorded in a natural environment, and subsequently examined their cortical complex sound representations. Cortical neurons became more responsive to dynamic and complex features of the complex sounds. In addition, more neurons were involved in representing the whole set of complex sounds, but fewer neurons actually responded to each individual sound. Cortical responses to different renderings of the same song motif were more similar, and responses to sounds of different motifs became more distinctive, indicating that cortical neurons were more selective to the defining features of the experienced sounds. These effects lead to better neural discrimination/recognition of the experienced complex sounds.

**9:20**

**3aAB5. From sounds to meaning: Neural representation of calls in the avian auditory cortex.** Julie E. Elie and Frederic E. Theunissen (Psych. Dept. & Helen Wills Neurosci. Inst., Univ. of California, Berkeley, 3200 Tolman Hall, Berkeley, CA 94720, julie.elie@berkeley.edu)

Understanding how the brain extracts meaning from communication sounds is a central question in auditory research. Communication sounds distinguish themselves not only by their acoustical properties but also by their information content. To study how the auditory system could differentially treat signals that have different social meanings, we investigated in zebra finches (*Taeniopygia guttata*) the perception of vocalizations that are used in clearly distinct contexts. We first generated a vocalization library containing the entire repertoire of female and male zebra finches. We then investigated the neural representations of these social calls in primary and secondary auditory areas. Using both simple measures of spike rate and optimal decoding methods based on entire PSTH, we indentified 24% of 1400 single units selective for the different call categories. To further understand how neurons could process sounds to generate selective responses, we compared models of the neural response based on the acoustic properties of sounds and/or on the semantic values of calls. We found neurons that were very well explained by the simple semantic model. Combining these results with the anatomical properties of cells (positions and spike shapes) gives new insight into the neural representation of meaningful stimuli in the avian auditory neural network.

**9:40**

**3aAB6. Cortical representation of complex spectrotemporal features in songbirds.** Gunsoo Kim, Helen McLendon (Physiol., UCSF, 675 Nelson Rising Ln., Rm. 521, San Francisco, CA 94143, gkim@phy.ucsf.edu), and Allison Doupe (Psychiatry, UCSF, San Francisco, CA)

Mechanistic understanding of how the auditory cortex processes complex communication signals remains a challenge. With their rich vocal communication behaviors, songbirds can offer insights into this question. We investigated the neural representation of sound features in the cortical auditory areas of zebra finches. In the primary cortical area field L, our systematic mapping of spectrotemporal receptive fields revealed a highly organized representation in which sharpness of spectral and temporal tuning of sound is mapped along two separate anatomical axes. The clustering of temporally or spectrally selective neurons suggested that initial cortical filtering for basic perceptual qualities such as tempo and pitch occurs in a spatially organized and segregated manner. Moreover, using an information theoretic based technique, we are uncovering additional sound features beyond those represented by conventional spectrotemporal receptive fields. In field L, we find that many neurons encode a second feature that typically captures rapid spectral or temporal modulations overlapping the first feature. In the secondary auditory area CM, a major target of field L, we are discovering an emergent sensitivity to frequency stacks, prevalent in zebra finch vocalizations. Together, our data show a systematic and hierarchical mapping of sound features onto songbird cortical neurons.

**10:00–10:20 Break**

**10:20**

**3aAB7. Neural encoding of learned auditory feedback statistics in avian vocal-motor circuitry.** Kristofer E. Bouchard (Neurosurgery, UCSF, San Francisco, CA) and Michael S. Brainard (Physiol., UCSF, 675 Nelson Rising Ln., San Francisco, CA 94158-0444, msb@phy.ucsf.edu)

Many complex behaviors are supported by neurons with both sensory and motor properties. During behavior such sensory-motor neurons experience the probabilistic associations of pre-motor activity for current actions with feedback from previous actions. Consequently, these associations might become encoded through Hebbian mechanisms. To investigate this possibility, we measured whether and how auditory-motor neurons in the vocal premotor nucleus HVC of songbirds encode the relative probabilities that different syllable sequences were produced during singing. We recorded from HVC while playing pseudo-randomly sequenced syllables from the bird's repertoire and found that auditory responses to syllables were positively modulated by the conditional probability that preceding sequences were produced during singing. Moreover, responses integrated over seven or more syllables, with the sign, gain, and temporal extent of integration depending strongly on probability. Our findings indicate that encoding of probabilistic associations between current and previous sounds may be a general principle of vocal-motor circuits.

10:40

**3aAB8. Local field potentials in the big brown bat inferior colliculus track the flow of objects moving past the bat.** James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu), Andrea M. Simmons (Cognit., Linguistic, and Psychol. Sci., Brown Univ., Providence, RI), Michaela Warnecke, and Jonathan R. Barchi (Neurosci., Brown Univ., Providence, RI)

Echolocating big brown bats orient and guide their flight in cluttered environments by following narrow corridors through the maze of surrounding obstacles. When flying in clutter, the bat aims its head, broadcast beam, and external ears to the front, to determine whether the upcoming path forward is safe to enter. This stabilizes the surroundings relative to flight direction to make the flow-field passing by the bat on the left and right stand out from the scene. To examine the capability of the bat's auditory system to follow the clutter as it moves past the bat, we recorded local field potentials (LFPs) from the inferior colliculus of anesthetized bats. We presented sounds that mimic FM biosonar broadcasts followed by clusters of echoes representing rows of objects moving past the bat. LFPs evoked by broadcast-echo sequences tracked the movements of the scene by registering arrays of echoes at latencies corresponding to echo delays. The limit for LFPs to follow multiple clutter objects is related to neuronal recovery times of several milliseconds. These data show that neural activity in the inferior colliculus represents dynamic changes in echo flow-fields that the bat would encounter in free flight. [Work supported by ONR and NSF.]

11:00

**3aAB9. Different forms of auditory-vocal feedback control in echolocating bats.** Walter Metzner (IBP, UCLA, 621 Ch. E. Young Dr. S., Los Angeles, CA 90095-1606, metzner@ucla.edu)

Auditory feedback from the animal's own voice is essential during bat echolocation: to optimize signal detection, bats continuously adjust various call parameters in response to changing echo signals. Horseshoe bats exhibit a particularly well-developed form of auditory feedback. Their echolocation pulses are dominated by a constant frequency component that matches the frequency range they hear best. To maintain echoes within this "auditory fovea," horseshoe bats constantly adjust their echolocation call frequency depending on the frequency of the returning echo signal. This Doppler-shift compensation behavior represents one of the most precise forms of sensory-motor feedback known. When examining the Lombard effect in horseshoe bats, we found that noise had different effects on call amplitude and frequency rises indicating different neural circuits and/or mechanisms underlying these changes. Both, amplitude and frequency rises were extremely fast and occurred in the first call uttered after noise onset, suggesting that, in contrast to Doppler-shift compensation, the Lombard effect did not require any auditory feedback. Bats also possess a large repertoire of communication calls, which differ greatly from those emitted during echolocation. We compared the variability of echolocation pulses and one common type of communication signal and found fundamentally different feedback mechanisms for echolocation and communication.



11:20

**3aAB10. Audiomotor activity in the superior colliculus of the big brown bat engaged in a natural, acoustic orientation task.** Melville J. Wohlgemuth and Cynthia F. Moss (Psych. and ISR, Univ. of Maryland, College Park, MD 20742, melville@umd.edu)

To accurately select and orient to a target, noisy, and multimodal sensory information about the target's location must be integrated into a coordinated set of orienting movements. At the hub of sensorimotor integration for species-specific orientation is the superior colliculus (SC), a midbrain structure receiving multimodal sensory inputs and projecting to premotor nuclei throughout the brainstem. Our research brings together behavioral and chronic neural recording data to examine auditory and premotor activity in the SC of the echolocating big brown bat as it performs a natural, goal-directed task. We trained bats to rest on a platform and track a tethered insect moved by a computerized stepper motor system. While the bat was tracking and capturing insects, single neuron activity was recorded across superficial, intermediate, and deep layers of the SC. Neural activity across the laminae of the SC signal auditory and pre-motor events: Echoes reflected from the sonar target evoked activity in superficial and intermediate layers, while premotor activity related to pinna, head, and vocal-motor behaviors was found at deeper recording sites. Collectively, the results of this study contribute to a deeper understanding of midbrain audiomotor activity in the context of natural goal-directed tasks.

11:40

**3aAB11. Neural processing of pressure and particle motion in central auditory pathways of larval bullfrogs.** Andrea Simmons and Victoria Flores (Cognit., Linguistic and Psychol. Sci., Brown Univ., Box 1821, Providence, RI 02912, Andrea_Simmons@brown.edu)

The metamorphic transition from an aquatic to a terrestrial milieu considerably impacts the functioning of the anuran auditory system. Neural responses to underwater particle motion produced by z-axis vibration and to pressure waves transmitted through the air/water interface can be recorded from the tadpole's dorsal medulla and torus semicircularis (auditory midbrain). Before metamorphic climax, these responses likely reflect stimulation of the saccule. Particle motion sensitivity in the dorsal medulla is stable in frequency range and sensitivity throughout larval development. In contrast, coding of both pressure waves and particle motion in the torus semicircularis is highly variable. There is a transient loss of pressure sensitivity in a short stage range ("deaf period") prior to metamorphic climax, correlated with the development of the middle ear. Robust responses to particle motion are seen in the torus semicircularis during late larval stages and throughout the "deaf period." During climax, however, these responses are considerably degraded or lost completely. We interpret this second "deaf period" to reflect central neural, rather than peripheral mechanical, effects, likely related to rerouting of afferent pathways.

## Session 3aAO

## Acoustical Oceanography: Munk Award Lecture

Andone C. Lavery, Chair
*Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536*

**Chair's Introduction—11:00**

### *Invited Paper*

**11:15**

**3pAO1. Ten years of seismic oceanography: Accomplishments and challenges.** W. Steven Holbrook (Geology and Geophys., Univ. of Wyoming, 1000 E. University Ave., Laramie, WY 82071, steveh@uwyo.edu)

"Seismic oceanography" (SO)—the use of low-frequency marine seismic reflection data to image thermohaline fine-structure in the water column—began in 2003, with the publication of a paper in Science. Over the past ten years, the nascent SO community has demonstrated that reflection seismology can image thermohaline fine structure, over large areas, from temperature contrasts in the ocean of only a few hundredths of a °C. The resulting images illuminate many diverse oceanic phenomena, including fronts, water mass boundaries, internal wave displacements, internal tide beams, eddies, turbulence, and lee waves. Beyond merely producing spectacular images of ocean structure, low-frequency reflections can be processed to produce quantitative estimates of sound speed (and thus ocean temperature), turbulence dissipation, and vertical mode structure over full ocean depths, as long as fine-structure reflections are present. Yet SO has failed to become a standard tool for physical oceanographers, partly due to disciplinary boundaries, and partly due to the perceived high expense of seismic data acquisition. I will present examples of the successes of SO and discuss approaches to meet the challenges to the adoption of SO as a commonly used technique to study physical oceanographic processes.

WEDNESDAY MORNING, 4 DECEMBER 2013 GOLDEN GATE 2/3, 7:55 A.M. TO 10:00 A.M.

## Session 3aBAa

## Biomedical Acoustics: Recent Advances in Therapeutic Ultrasound I

Lawrence Crum, Cochair
*Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105*

Kullervo Hynynen, Cochair
*Medical Biophys., Univ. of Toronto, Sunnybrook Health Sci. Ctr., Toronto, ON M4N 3M5, Canada*

**Chair's Introduction—7:55**

### *Invited Papers*

**8:00**

**3aBAa1. Advances in ultrasound methods for therapy.** Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Sunnybrook Health Sci. Ctr., Toronto, ON M4N 3M5, Canada, khynynen@sri.utoronto.ca), Alison Burgess, Meaghan M. O'Reilly (Physical Sci. Platform, Sunnybrook Res. Inst., Toronto, ON, Canada), Ryan Alkins, Daniel Pajek, Nicholas Ellens, and Alec Hughes (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Focused ultrasound has been shown to be the only method that allows noninvasive thermal coagulation of tissues and recently this potential has been explored for noninvasive image-guided drug delivery. In this presentation, the advances in ultrasound phased array technology for well controlled energy delivery will be discussed. In addition, some of the recent preclinical results for the treatments of brain tumors, stroke, and Alzheimer's disease will be reviewed. As conclusion, the advances in the image-guided focused ultrasound for the treatment of disease has been rapid and the future potential appears very promising.

**8:20**

**3aBAa2. Catheter-based and endoluminal ultrasound applicators for magnetic resonance image-guided thermal therapy of pancreatic cancer: Preliminary investigations.** Chris Diederich, Vasant Salgaonkar, Punit Prakash, Matt Adams, Serena Scott, Peter Jones, Daniel Hensley, Henry Chen (Radiation Oncology, UCSF, 1600 Divisadero St., Ste. H1031, San Francisco, CA 94143-1708, cdiederich@radonc.ucsf.edu), Juan Plata, Andrew Holbrook, Kim Butts Pauly, and Graham Sommer (Radiology Dept., Stanford Univ., Stanford, CA)

Ultrasound devices are being investigated for endoluminal and intraductal access for targeted thermal ablation or hyperthermia of pancreas under MR guidance and temperature monitoring. Simulations using patient-specific 3D models were developed for applicator design and development of treatment delivery strategies. MR-compatible devices were constructed for endoluminal (3-5 MHz planar or lightly focused rectangular elements, 12-mm OD assembly, expandable balloon), transgastric interstitial and intraductal (6-8 MHz multi-sectored tubular elements, 2-mm catheter) deployment. Micro-coils were integrated for active MR tracking of position and alignment. The proof-of-concept devices were tested in phantoms, *ex vivo* tissues, cadaveric porcine models, and *in vivo* animal models under 3T MR temperature imaging (MRTI). Results indicate endoluminal devices could ablate 2-2.5 cm depth from gastric wall for tumors of the pancreatic head, and multi-sectored tubular intraductal and interstitial applicators could ablate 2.3-3.4 cm diameter targets with directional control. Intraductal applicators could produce effective hyperthermia (>40 C) extending 15 mm radial. Customized tracking sequences could be used to locate 3D position of the applicators. Endoluminal, interstitial, and intraductal ultrasound applicators show promise for ablation or hyperthermia of pancreatic tumors. MR guidance can be employed for positioning these devices with active tracking coils and real time temperature monitoring. (NIH-P01CA159992.)

**8:40**

**3aBAa3. Targeted drug delivery to the brain and brain tumors using focused ultrasound and microbubbles.** Nathan McDannold (Radiology, Brigham and Women's Hospital, 75 Francis St., Boston, MA, njm@bwh.harvard.edu)

The physiology of the vasculature in the central nervous system (CNS), which includes the blood-brain barrier (BBB) and other factors, severely limits the delivery of most drugs to the brain and to brain tumors. Focused ultrasound (FUS), when combined with circulating microbubbles, is a noninvasive method to locally and transiently disrupt the BBB at discrete targets and enhance delivery across the "blood-tumor barrier." This talk aims to provide insight on the current status of this unique drug delivery technique, experience with it in preclinical models, and its potential for clinical translation. In particular, methods to monitor the procedure using acoustic receivers and the feasibility of controlling and predicting drug deposition will be reviewed. If this method, which offers a flexible means to target therapeutics to desired points or volumes in the brain, can be translated to the use in humans, it can enable the use of the whole arsenal of drugs in the CNS that are currently prevented by the BBB.



**9:00**

**3aBAa4. Mechanism, monitoring, and drug delivery of the ultrasound-induced blood-brain barrier opening.** Elisa Konofagou (Biomedical Eng., Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

Worldwide, neurodegenerative diseases account for more than 20 million patients. Aging greatly increases the risk of neurodegenerative disease while the average age of Americans is steadily increasing. Numerous small- and large-molecule drugs have been developed for treatment of neurodegenerative diseases but with mixed success. This is mainly because, when administered systemically *in vivo*, the blood-brain barrier (BBB) inhibits their delivery to the regions affected. Safe and localized opening of the BBB has been proven to present a significant challenge. Focused ultrasound (FUS) in conjunction with microbubbles remains the sole technique that can induce localized BBB opening noninvasively, selectively, and transiently. Over the past few years, our group has been able to unveil several aspects of the technology in order to (a) unveil the physical mechanism of opening, (b) maintain safety, (c) establish a non-MRI type of monitoring technique, (d) control the volume and permeability of opening through the microbubble used, (e) demonstrate large animal feasibility, and (f) determine the range of molecular sizes delivered. We have also shown that neurotrophic agents are capable of triggering downstream effects into the neuronal nucleus through the induced opening. All the aforementioned aspects including initial drug efficacy findings in large animals will be discussed.

**9:20**

**3aBAa5. High-intensity focused ultrasound treatment of prostate cancer.** Narendra T. Sanghvi (R & D, SonaCare Medical, 4000 Pendleton Way, Indianapolis, IN 46226, narensanghvi@sonacaremedical.com)

In the last decade, over 40,000 prostate cancer patients have been treated by HIFU systems in over 30 countries. These treatments have been conducted using two ultrasound image guided hifu devices—Ablatherm (EDAP, Lyon, France) and Sonablate® 500 (Focus Surgery, Inc., Indianapolis, IN). In addition, there is a shift in the management of prostate cancer from whole gland radical prostatectomy and radiation to focal treatment of prostate cancer. The focal treatment is guided by meticulous pretreatment imaging with multiparametric MRI to accurately localize the index lesion. The MRI images are used to render 3D deformable model of the prostate gland and provide fusion of US and MRI to guide HIFU treatment resulting in reduced complications of rectal fistula, erectile dysfunction, and urinary incontinence. The results of the clinical studies indicate that patients with recurrent cancer post radiation can benefit from HIFU treatment. Both these devices are marketed in many countries and recently have submitted PMA applications to the FDA to receive clearance to market in the United States. Long term clinical results and status of HIFU devices will be presented.

9:40

**3aBAa6. Ultrasound-based neurostimulation in the mouse model.** Kim Butts Pauly (Radiology, Stanford Univ., Lucas Ctr., 1201 Welch Rd., Stanford, CA 94305, kbpauly@stanford.edu), Randy King, Patrick Ye (BioEng., Stanford Univ., Stanford, CA), and Julian Brown (Neurobiology, Stanford Univ., Stanford, CA)

Ultrasound-based neurostimulation would be a useful tool prior to MR-guided focused ultrasound treatments in the brain. In this work, we report on our studies on ultrasound-based neurostimulation in the mouse model. We define the success rate as the ratio of the number of positive EMG responses to the number of sonications. A single element ultrasound transducer with a center frequency of 500 kHz was applied to the mouse head via a coupling column and coupling gel on the mouse head. EMG electrodes were placed in the mouse neck and tail muscles to measure contraction of the relevant muscles as the ultrasound transducer is moved across the mouse head. The success rate increases with ultrasound intensity or with ultrasound duration, following a sigmoidal curve. As the ultrasound frequency is increased, the ultrasound intensity must be increased for the same success rate. Movement of the ultrasound transducer across the brain changes the response in the relevant EMG systems such that the neck EMG response is stronger when the transducer is more rostrally placed, while the tail EMG response is stronger when the transducer is more caudally placed. Our findings present evidence for selective targeting in the mouse model of ultrasound-based neurostimulation.

WEDNESDAY MORNING, 4 DECEMBER 2013 GOLDEN GATE 2/3, 10:25 A.M. TO 11:30 A.M.

## Session 3aBAb

## Biomedical Acoustics: Distinguished Lecture: The Use of Magnetic Resonance-Guided High Intensity Focused Ultrasound to Treat Essential Tremor (ET)

Lawrence Crum, Cochair
*Appl. Phys. Lab., Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105*

Kullervo Hynynen, Cochair
*Medical Biophys., Univ. of Toronto, Sunnybrook Health Sci. Ctr., Toronto, ON M4N 3M5, Canada*

**Chair's Introduction—10:25**

***Invited Paper***

10:30

**3aBAb1. The use of magnetic resonance-guided high intensity focused ultrasound to treat essential tremor.** William J. Elias, Diane Huss (Neurosurgery, Univ. of Virginia, Box 800212, UVA HSC, Charlottesville, VA 22908, wje4r@virginia.edu), Tiffini Voss (Neurology, Univ. of Virginia, Charlottesville, VA), Johanna Loomba, Mohamad Khaled, Robert Frysinger (Neurosurgery, Univ. of Virginia, Charlottesville, VA), Scott Sperling, Scott Wylie (Neurology, Univ. of Virginia, Charlottesville, VA), Stephen Monteith (Neurosurgery, Univ. of Virginia, Charlottesville, VA), Jason Druzgal (Neuroradiology, Univ. of Virginia, Charlottesville, VA), Binit Shah, Madaline Harrison (Neurology, Univ. of Virginia, Charlottesville, VA), and Max Wintermark (Neuroradiology, Univ. of Virginia, Charlottesville, VA)

Advances in ultrasound transducer technology have enabled for transcranial sonication with energy levels adequate to achieve tissue ablation. With MR-guidance and monitoring, precise lesioning is now possible of deep brain targets such as the thalamus and basal ganglia so that stereotactic lesioning is being reconsidered for the treatment of movement disorders. In this phase 1 clinical trial, we investigate the feasibility and safety of MRgFUS for performing a unilateral thalamotomy for medication-refractory essential tremor (ET). According to an FDA-approved protocol, 15 patients with medication-resistant ET underwent unilateral MRgFUS lesioning of the thalamus for dominant limb tremor. Intraprocedural monitoring was conducted with each incremental sonication using MR thermometry and clinical examination. Neurological assessments, validated tremor ratings, MRI, and quality of life data were recorded preoperatively and during a year post treatment. Adverse events were recorded throughout the study duration. Accurate thalamic lesioning was achieved in all cases. Dominant limb tremor subscores improved by nearly 75% while ipsilateral limb tremor was unchanged. Functional activities and quality of life measures improved significantly. Refining of the thalamic target was possible in five cases with subthreshold sonications. Serial MR imaging defined the evolution of the lesioning process.

# Session 3aEA

## Engineering Acoustics: Non-Traditional Electro-Acoustic Transducer Design I

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

**Chair's Introduction—8:30**

### *Invited Papers*

**8:35**

**3aEA1. Thermophone projectors using nanostructure materials.** Benjamin Dzikowicz, Jeffrey W. Baldwin, and James F. Tressler (Code 7130, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, ben.dzikowicz@nrl.navy.mil)

Thermophone transducers fabricated from new nanoscale materials hold the promise of a new transducer technology for the Navy with no moving parts that operate over a broad frequency range and can be designed to be lighter and thinner than competing technologies. This potentially makes them ideal for use as high performance conformal projectors on autonomous underwater vehicles, submarines, and other small craft. Although thermophone devices have been understood for nearly a century, [Phys. Rev. **10**, 22 (1917)], new nanostructure materials with extremely low heat capacities per surface area have recently become available which have the potential of greatly increasing their efficiency. Thermodynamic models show that certain surfaces, gasses, and enclosures will increase the acoustic efficiency. However each of these modifications of the base design has drawbacks as well. These will be discussed from a theoretical standpoint and results from laboratory testing will help to verify these hypotheses. [Work supported by NRL.]



**8:55**

**3aEA2. Thermoacoustic sound projectors using carbon nanotubes and other nanostructures.** Ali E. Aliev and Ray H. Baughman (Alan G. MacDiarmid NanoTech Inst., Univ. of Texas at Dallas, P.O. Box 830688, BE 26, Richardson, TX 75083, Ali.Aliev@utdallas.edu)

The application of solid-state fabricated carbon nanotube sheets as thermoacoustic (TA) projectors is extended from air to underwater applications. Due to non-resonant sound generation, the emission spectrum of nanotube sheets in air or underwater varies smoothly over a wide frequency range, 1-$10^5$ Hz. Encapsulating the nanotube sheet projectors using inert gases with low heat capacity provided attractive performance at needed low frequencies, as well as a realized energy conversion efficiency in air of 0.2% and 1.5% underwater, which can be enhanced by further increasing the modulation temperature. We suggest enhancement of sound generation efficiency of encapsulated device by using high quality resonant acoustical windows and modulation of high frequency carrier current with a low frequency resonant envelope. Applications of TA projectors for high power sonar arrays and transparent flexible loudspeakers will be discussed. Finally, the alternative nanostructures for excitation of thermoacoustic sound waves will be surveyed. [We gratefully acknowledge support by Office of Naval Research grant N00014-13-1-0180.]

**9:15**

**3aEA3. First look: Acoustic calibration of carbon nanotube transducers.** Dehua Huang and Thomas R. Howarth (NAVSEA Div. Newport, 1176 Howell St., B1346 R404A, Newport, RI 02841, thomas.howarth@navy.mil)

Material researchers at the University of Texas at Dallas (UT-D) have recently been reporting on the development of underwater acoustic carbon nanotube (CNT) yarn sheets capable of high acoustic output at low frequencies with broad bandwidth. The principle transduction mechanism for their approach is through thermal acoustic means as opposed to conventional underwater transducers that utilize electromechanical vibrations. This presentation will begin with an overview of the CNTs including the design of a first generation packaging technique that was incorporated for the fabrication of three prototypes. The prototypes were acoustically calibrated in April 2013 at the US Navy acoustic calibration facility in Okahumpka, Florida. The presentation will include measured unbiased and biased transmitting voltage responses (TVRs) and directivity patterns over a two and a half decade band. Final discussions will include ongoing research directions for further development. [Work supported by NAVSEA Division Newport.]

**9:35**

**3aEA4. Carbon nanotube thermoacoustic projectors for undersea vehicles.** Michael R. Zarnetske and John B. Blottman (Sensor and Sonar Syst., Naval Undersea Warfare Ctr., 1176 Howell St., B1170/R109, Newport, RI 02841, michael.zarnetske@navy.mil)

Renewed interest in the thermophone has developed with the recently demonstrated capability to manufacture carbon nanotube thin films and capacity to emit sound through the thermoacoustic effect. High fidelity broadband sound generation is attributed to the ultra-low heat capacity and low thermal inertia of these films. Motivated by the need for low-frequency, broad-bandwidth, compact sonar projectors to be embedded in the hull of unmanned sea vehicles or in the outer coating of a surface combatant or submarine as a conformal

array, a team of researchers from Virginia Tech, The University of Texas at Dallas, and the Naval Undersea Warfare Center are evaluating these novel materials and devices both theoretically and experimentally. Analytical and numerical simulations support mechanical, thermal and acoustic experimentation. Correlated results will be presented. [Work supported by Office of Naval Research, code 321MS.]

**9:55**

**3aEA5. Alternative tonpilz and bender transducer designs.** John Butler (Image Acoust., Inc, 97 Elm St., Cohasset, MA 02025, jbutler@imageacoustics.com)

Tonpilz transducer designs with half-wavelength water-sized radiating-pistons are commonly used in SONAR arrays. And here the tonpilz piston normally radiates in the broadside direction with reduced output as the array is steered to the end-fire direction. Bender transducer designs, used in low frequency applications, can take the form of a dipole transducer which, as a result of partial self-cancellation, can lead to a low source level output. We present two alternative transducer designs, which solve these problems. In the case of the tonpilz array, a cylindrical-shaped leveraged-transducer design with one-half water wavelength size and modal performance is proposed. This transducer is shown to operate in the first three modes allowing the formation of an element beam that may be steered in the general directions the array is steered, with full output in a single end-fire direction. In the case of the dipole transducer, advantage is taken of the strong near-field dipole acoustic pressure which is used to energize a nearby compliant parasitic resonator yielding a dominant monopole source of greater output. [Work supported in part by ONR.]

**10:15–10:30 Break**

## *Contributed Papers*

**10:30**

**3aEA6. Multi-degree-of-freedom model of 32(1)-mode cylindrical transducer with inactive elements.** Nicholas Joseph and Michael R. Haberman (Mech. Eng., Appl. Res. Labs., The Univ. of Texas at Austin, 901 East 40th St., Apt. 301, Austin, TX 78751, nickjjoseph@gmail.com)

Piezoelectric transducers with cylindrical geometry are often designed to operate in a radial "breathing" mode. In order to tune their performance in a cost effective way, cylinders can be constructed of alternating active (piezoelectric) and inactive (non-piezoelectric) staves. Existing lumped parameter models for such a ring are based on effective piezoelectric properties of the composite ring which reduce the system to a single degree of freedom corresponding to the breathing motion [Butler, J. Acoust. Soc. Am., **59**(2), 480-482, (1976)]. Unfortunately, if the length of the staves is a sufficiently large percentage of the circumference, the transducer may demonstrate a detrimental higher frequency resonance within the desired bandwidth of operation. This parasitic resonance results from bending motion of the staves and can significantly decrease the radiated acoustic pressure and generate distortion. This work presents a multiple-degree-of-freedom lumped parameter model that captures both the breathing and bending resonances of the transducer and provides a more accurate prediction of its effective coupling coefficient. Results are compared with a one-degree-of-freedom model, finite element models, and experimental data. Modifications to account for internal volumes and nonlinear effects are also presented and discussed.

**10:45**

**3aEA7. Design optimization of a piezoelectric microphone with in-plane directivity.** Michael L. Kuntzman, Nishshanka N. Hewa-Kasakarage, Donghwan Kim (Elec. and Comput. Eng., The Univ. of Texas at Austin, 2501 Speedway, Stop C0803, Austin, TX 78712, mlkuntzman@gmail.com), Alex Rocha (Microelectronics Res. Ctr., The Univ. of Texas at Austin, Austin, TX), and Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

A piezoelectric micromachined microphone with in-plane directivity has been recently introduced [Appl. Phys. Lett. **102**, 054109 (2013)]. The work is inspired by a design introduced by Miles *et al.* [J. Acoust. Soc. Am. **125**(4), 2009], which was, in-turn, inspired by the hearing mechanism of a particular type of parasitoid fly. A rocking structure pivots about a rotational hinge in response to in-plane pressure gradients, and the rocking motion is read by springs attached to the end of the rocking 'teeter-totter' structure, with the springs themselves employing thin piezoelectric films, which operate in a 3-1 mode. Prototypes have been fabricated that employ rocking structures 1 mm x 2 mm in size and functionality has been verified via directivity measurements performed in an anechoic chamber. This presentation will focus on exploring the design space of this sensor, which is accomplished with a hybrid model based on FEA and network models. Designs which maximize SNR are presented, and anticipated microfabrication challenges of these designs are highlighted.

**11:00**

**3aEA8. Tuning a combustive sound source to meet experimental needs.** Andrew R. McNeese, Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. 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 spark ignited. Upon ignition, the combustive mixture is converted into high temperature combustion byproducts, which expand and ultimately collapse to smaller volume than before ignition. Acoustic pulses are radiated by the bubble activity. The CSS can be used as a source for array calibration, propagation measurements, bottom characterizations, and sea floor seismic testing. Current environmental regulations and varying experimental needs require a tunable source that allows users to easily alter the source level, bandwidth, center frequency, and signal duration. Present efforts have focused on designing and testing a variety of devices that alter the resultant bubble activity to tune the radiated signals to meet various experimental needs. A new combustion chamber and gas exit ports have been constructed and tested in tank experiments. The results show that the resultant acoustic pulses can indeed be varied, and that tone bursts can be created. Discussions show how the device was tuned to meet specific needs for a particular application. [Work supported by ARL:UT Austin.]

**11:15**

**3aEA9. An ultrasonic actuator working under cryogenic and vacuum circumstance.** Zhuzi Chen, Yu Chen, Tieying Zhou, and Deyong Fu (Dept. of Phys., Tsinghua Univ., Beijing 100084, China, chenyu@tsinghua.edu.cn)

In this work, we present an ultrasonic actuator that can work under cryogenic vacuum environment. It can be used for adjusting distance between capacitor electrodes in high temperature superconductor filter (HTSF) to tune its pass-band. The actuator is an single crystal chips driven nut-type ultrasonic motor, which can work under cryogenic vacuum conditions. The stator of a nut-type ultrasonic motor is a nut-shaped octagon with internal thread, which matches with the rotor external thread and a bottom at one end as fixing base. Piezoelectric chips are glued to the sides of the octagon to generate a traveling wave along the circumference. Vibration of the stator

drives the rotor to rotate via friction between thread interfaces and the thread drives the rotor to move along the axis direction. The actuator that we developed can easily acquire micrometer positioning accuracy, which enables it to tune the pass-band of a HTSF effectively. The motor was optimized with FEM harmonic response analysis. The mechanical characteristics and stepping precision of the prototype ultrasonic motor have been measured and discussed. [Work supported by NSFC.]

**11:30**

**3aEA10. Underwater low frequency acoustic projector based on a musical instrument design.** Andrew A. Acquaviva and Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ.,University Park, PA, aaa261@psu.edu)

An electroacoustic projector that is small compared to the radiated wavelength presents a significant design challenge because its radiation resistance is small and the radiation impedance is highly reactive. Practical designs are often limited by high dynamic strain and high reactive drive currents. Wind musical instruments are also small compared to their lowest playing frequencies. However, wind musical instruments are not limited by the low radiation impedance. They use the low radiation impedance as a part of the regeneration mechanism that sustains the oscillation. This paper examines the hypothesis that an underwater projector designed to operate in water in the manner that a wind instrument operates in air may provide performance that is competitive with conventional electroacoustic low frequency acoustic projectors.

WEDNESDAY MORNING, 4 DECEMBER 2013 EAST LOUNGE, 10:00 A.M. TO 12:00 NOON

## Session 3aEDa

## Education in Acoustics: Undergraduate Research Exposition Poster Session

Mardi C. Hastings, Cochair
*Georgia Inst. of Technol., George W. Woodruff School of Mech. Eng., 801 Ferst Dr., Atlanta, GA 30332-0405*

Preston S. Wilson, Cochair
*Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292*



### *Contributed Papers*

All posters will be on display and all authors will be at their posters from 10:00 a.m. to 12.00 noon.

**3aEDa1. Experimental and numerical analysis of the effects of depth, diameter, and tension in musical drumhead coupling.** Benjamin Boe and Randy Worland (Phys. Dept., Univ. of Puget Sound, 1500 N Waner, Tacoma, WA 98416, bboe@pugetsound.edu)

The low frequency vibrations of two-headed musical drums are known to couple. However, little is known regarding the factors that determine the degree of coupling at higher frequencies. In this study, the effects that depth, diameter, and head tension have on the tendency of the drumheads to couple are investigated. Commercial finite element software was used to model a wide range of drums and to identify trends of coupling according to these factors. The numerical results were used to guide which parameters should be tested in the lab. Experimentally, two oppositely facing Electronic Speckle-Pattern Interferometry systems were used to optically view the simultaneous vibrations of both heads of a drum. Several snare and tom tom drums with different diameters, depths, and tensions were observed. To closely analyze the effect of drum depth a tom tom was modified so a range of depths from 1.5'' to 40'' could be tested while keeping the diameter and tension constant. The optical and numerical data are used to illustrate trends in the coupling of musical drumheads.

**3aEDa2. The real world transmission loss chamber: A work in progress.** Jay Bliefnick, Andrew M. Hulva, Dominique J. Chéenne (Audio Arts & Acoust., Columbia College Chicago, 5001, Apt. 1S, Schiller Park, IL 60176, jay.bliefnick@loop.colum.edu)

A "real world" transmission loss (RWTL) chamber was recently added to the undergraduate acoustics program at Columbia College Chicago. It aims to demonstrate concepts of transmission loss to students and to provide a "less-than-ideal" environment for construction evaluation prior to certified tests with full-size samples. Each side of the chamber can be used as either "send" or "receive," and both speaker and microphone placements are infinitely variable. The noise floor can also be adjusted to illustrate potential issues in field tests. Measurements are taken simultaneously from multiple positions, then averaged to yield TL values and an overall isolation rating. Given its construction, the chamber is not expected to produce results as per existing standards as its small size results in modal effects and non-diffuse conditions on both sides of the tested partition. This study's first goal was to better understand the performance limitations of the RWTL chamber by conducting a thorough evaluation of its maximum TL and modal properties. The second goal was to optimize the chamber's testing methodology in order to more closely reflect certified laboratory results; this was done by utilizing recently obtained certified data from a door manufacturer who tested reduced-sized samples in the RWTL chamber.

**3aEDa3. Just noticeable differential estimation of source-receptor dislocations in the auralization process.** Bernardo Murta (Undergrad. Program Acoust. Eng., DECC-CT-UFSM, Federal Univ. of Santa Maria, Rua da Passagem, 111, Belo Horizonte, Minas Gerais 30220-390, Brazil, be.murta@gmail.com), Jessica J. Lins, and Stephan Paul (Undergrad. Program Acoust. Eng., DECC-CT-UFSM, Federal Univ. of Santa Maria, Santa Maria, Brazil)

The precision of source-receptor transfer functions is of importance to provide reliable and ecologically valid results in auralization. A test using paired comparison was developed to estimate the overall just noticeable differential (jnd) in signals obtained from the convolution of music with slightly varying source-receptor transfer functions. Subjects from 19 to 28 years, both male and female, were selected to participate comparing 24 sets of two sounds in each battery and answering if within each pair differences are noticeable. The signals were obtained by convolution of popular rock music with 12 s of duration with impulsive responses of different source-receptor positions in a scaled room and were divided in three batteries with different dislocation directions. To check subjects reliability pairs containing the same signals were also evaluated. The other 20 pairs were made by combinations of signals obtained by the FRF of the reference position and the FRFs of positions dislocated up to 10 cm from the reference point in a random manner. By analyzing the psychometric function, it was found that the jnd was reached when the receiver position was varied about 3 to 4 cm from the reference position. Further tests will be done to clarify details.

**3aEDa4. Comparing the time variance of orchestral instrument directivities in Mozart's symphony in G-minor: First movement.** Kristin Hanna and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 107 Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0816, khanna@unomaha.edu)

Multiple channel anechoic recordings of the musical instrument parts in Mozart's Symphony in G-Minor (First Movement) are analyzed to study how the directivity patterns of each instrument varies with time. Static directivity patterns are well-documented for many musical instruments, but studies on how their directional patterns vary with time are not as common. Changing directional patterns in time, however, have been found to impact the realism of room acoustic computer modeling simulations. Previous work at the University of Nebraska has suggested a method for studying the time variance of musical instruments across a number of simultaneously recorded channels in an anechoic chamber. The method involves time-windowing each channel and analyzing how the overall directivity index changes across time and frequency. Comparisons of results from some of the fourteen instruments included in this Mozart symphony are presented. [Work supported by a UNL Undergraduate Creative Activities and Research Experience Grant.]

**3aEDa5. An analysis of firefighter personal safety alarm effectiveness on the fire ground.** Kyle Ford, Mudeer Habeeb, Joelle Suits, Mustafa Abbasi (Dept. of Mech. Eng., Univ. of Texas at Austin, 5907 Trabadora Cove, Austin, TX 78759, kyleford@utexas.edu), Preston S. Wilson (Appl. Res. Lab. and Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX), and Ofodike Ezekoye (Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

For firefighters in the line of duty, the last line of defense and chance for rescue oftentimes relies on the effectiveness of their Personal Alert Safety System (PASS) devices. When activated, a PASS device emits an alarm signal to notify others that a firefighter is in distress. However, there have been notable instances where PASS devices have confused rescue personnel or created a more hazardous situation, for instance, when noise interference originating from other objects is involved. This research compiles data from various sources, for example, firefighter near miss reports and National Institute for Occupational Safety and Health (NIOSH) fatality reports, regarding PASS device effectiveness. The research will investigate the causes of confusion and danger as well as take a look at the situations where the device achieved its goal and was able to save a life. The implications of discovering how interfering noises can render PASS devices ineffective could save several lives in the future and ultimately lead to increased firefighter safety.

**3aEDa6. Correlation analysis of military aircraft jet noise.** Zachary Anderson, Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., 345 E 600 N F1, Provo, UT 84606, zachary-anderson@hotmail.com), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Correlation analysis is useful in extracting spatiotemporal relationships between signals and can be used to examine features of near-field jet noise for source properties. Characteristic correlation envelopes determined by Harker *et al.* [J. Acoust. Soc. Am. **133**, EL458 (2013)] can be used to relate correlation lengths to fine and large-scale turbulent structures. As an extension, cross-correlation shows spatial variation in jet noise and further reveals the transition between short (fine-scale) and long (large-scale) correlation lengths. These analyses are applied to a military jet dataset of a ground based linear microphone array positioned 11.6 m from the jet axis. Correlation analyses over multiple engine conditions and observation directions are reported. In particular, a maximum correlation coefficient greater than 0.5 exists over a range spanning multiple wavelengths in the region of greatest overall sound pressure level at military power. [Work supported by ONR.]

**3aEDa7. Autocorrelation analysis of lab-scale jets.** Kelly R. Martin, Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, kellymartin013@gmail.com), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Ashville, NC)

Autocorrelation (AC) analysis is useful in examining temporal relationships in a waveform and can be used to provide insight into properties of jet noise. Using techniques developed by Harker *et al.* [J. Acoust. Soc. Am. **133**, EL458 (2013)] for full-scale jet data, AC analysis has been applied to unheated, laboratory-scale jet noise data. To more consistently compare the AC at various locations around the jet, it is important to account for the spatial variation in the spectrum by scaling with the peak frequency. In addition to this frequency scaling, the spatiotemporal variations in the autocorrelation are more plainly seen when an envelope function is applied. Calculated AC envelope functions from measured data are compared with theoretical curves for fine and large-scale jet noise radiation. Results are compared against those from a full-scale, military jet aircraft. [Work supported by ONR.]

**3aEDa8. A rapid computational method to investigate the directivities of quasi-omnidirectional sources of sound.** Jeshua H. Mortensen and Timothy W. Leishman (Phys. and Astronomy, Brigham Young Univ., 765 N 400 E, Provo, UT 84606, meako490@gmail.com)

While dodecahedron loudspeakers are widely used in acoustical measurements as quasi-omnidirectional sources of sound, other multiple-driver configurations may also be used for this purpose. Previous experimental work has shown that loudspeakers with higher-order Platonic solid geometries tend to produce higher omnidirectional cutoff frequencies than their lower-order counterparts. However, as their radiated fields transition from omnidirectional to multidirectional at higher frequencies, their directivities may or may not be closer to the omnidirectional ideal. Additional testing has been required to better understand the effects, but it has been cumbersome because of the difficulty of constructing and measuring many modified loudspeakers. This poster presents a practical method to estimate the directional characteristics of multiple-driver sources based on spherical enclosure geometries and the use of common mathematical software such as MATLAB. It enables one to easily and rapidly predict the directivity patterns of these sources and the effects of altered driver diameters, positions, numbers, vibrational patterns, and enclosure volumes. The method is shown to produce several interesting results that are validated by the boundary element method and experimental measurements.

**3aEDa9. Evaluation of a small variable-acoustics chamber for speech accommodation research.** Matthew F. Calton, Timothy W. Leishman (Phys. and Astronomy, Brigham Young Univ., 266 N. 300 E. #26, Provo, UT 84606, mattcalton@gmail.com), and Eric J. Hunter (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT)

Many studies have been conducted over the years to explore speech in rooms and its intelligibility to listeners. Speech accommodation by talkers is another developing field in speech and architectural acoustics. In some occupations, a talker's voice is used nearly continuously throughout the workday. Acoustical conditions in the workplace can significantly affect vocal effort and the health and longevity of the vocal folds. Experimental resources are needed to better understand these conditions and how they may be optimized for the well-being of talkers. The present study investigates the range of acoustical conditions that may be established in a small variable-acoustics chamber for this type of research. The chamber is characterized using many pertinent room-acoustics parameters. Volunteers read passages in the chamber with no visual cues to impact their perception of its

changing acoustical treatments. Various measurements were made to establish relationships between the room conditions and vocal efforts.

**3aEDa10. Assessing the effectiveness of geometrically modified pyramidal diffusers: Scattering coefficient measurements.** Ariana F. Sharma and David T. Bradley (Phys. + Astronomy, Vassar College, 124 Raymond Ave., Poughkeepsie, NY 12604, arsharma@vassar.edu)

A diffuser is a surface with a non-planar geometry used in acoustically sensitive spaces to help mitigate unwanted effects from strong reflections such as echoes and focusing. Although a variety of diffuser surface geometries exist, new designs are constantly being generated for use in specific real-world projects in an effort to expand the aesthetic options available to the architect and acoustical consultant. The effectiveness of these new designs must be determined as part of the responsible acoustical design process. In the current project, the acoustic behavior of surfaces with a pyramidal base pattern has been measured according to standard ISO 17497-1. In particular, the standard outlines the measurement of the scattering coefficient, a quantifier of how much energy has been reflected away from the specular direction. This coefficient gives a general indication of the diffusing effectiveness of the surface and is useful in computational acoustics room modeling. The surface pyramidal base pattern has been varied to create an array of surfaces in an effort to find an optimal combination of geometric input parameters. Certain measurement conditions prescribed in the ISO standard were also varied to determine their effect on measurement accuracy. Results and analysis will be presented.

**3aEDa11. Simulated acoustical environments for the evaluation of vocal effort.** Jennifer Whiting (Brigham Young Univ., 2011 South 1175 East, Bountiful, UT 84010, lundjenny@comcast.net), Timothy Leishman (Brigham Young Univ., Provo, UT), and Eric Hunter (National Ctr. for Voice and Speech, Univ. of Utah, Salt Lake City, UT)

Realistic simulations of acoustical environments allow researchers to quickly manipulate the auditory experiences of human subjects. For studies investigating human speakers' perceptions of their own voices, a mixed-reality environment is most suitable. In this work, we have developed a system to create low-latency real-time convolutions of speakers' voices with simulated room impulse responses at their ears. The latter are based in part on measured voice directivity patterns. The simulated rooms included classrooms, lecture halls, and auditoria. They were all simulated for human speakers within an anechoic chamber. We also added realistic noise to the simulations, including chatter and other ambient effects, and measured the subjects' vocal efforts. The ultimate aim of the simulation and measurement system is to assess teachers' vocal efforts in classrooms and other settings with easily controlled acoustical conditions. However, the setup may also be easily adapted to other studies for speech or music.

**3aEDa12. Investigating tonal spaces using an extension of VoiceSauce voice analysis software.** Kate Silverstein and Kristine M. Yu (Linguist, Univ. of Massachusetts, 181 Presidents Dr, Amherst, MA 01003, ksilvers@student.umass.edu)

We extended VOICESAUCE (Shue, Keating, and Vicenik, 2009), a MATLAB application which provides automated voice measurements over time from audio recordings, to include utilities for command line processing and testing. The command line utilities allow users to access core VOICESAUCE functionality, including batch processing of wave (*.wav) files and parameter manipulation, independently of a graphical user interface. The testing framework provides an automated process for tracking and measuring the effects of manipulating parameter settings across runs. In addition, we modified VOICESAUCE to be compatible with Octave and ported it to Python in order to facilitate use and development from a wider community. We use this software to compare the inclusion of phonation measures in the set of voice source parameters against f0 alone across White Hmong and Cantonese. Phonation, specifically breathy voice, plays a perceptual role in tone identification in both languages; however, in White Hmong, breathy voice is a necessary cue for accurate tonal identification (Garellek *et al.*, 2012) whereas in Cantonese, phonation may facilitate perception but is not critical (Yu, 2011).



WEDNESDAY MORNING, 4 DECEMBER 2013 CONTINENTAL 5, 10:00 A.M. TO 2:00 P.M.

## Session 3aEDb

## Education in Acoustics: Hands-On Acoustics Demonstrations for Middle- and High-School Students

David T. Bradley, Cochair
*Phys. + Astronomy, Vassar College, 124 Raymond Ave., #745, Poughkeepsie, NY 12604*

Andrew C. Morrison, Cochair
*Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431*

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomenon. In this session "Hands-On" demonstrations will be set-up for a group of middle school students from the San Francisco area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students' educational development and is part of the larger "Listen Up" education outreach effort by the ASA. Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Any acousticians wanting to participate in this fun event should email David T. Bradley (dabradley@vassar.edu) or Andrew C. H. Morrison (amorrison@jjc.edu).

## Session 3aNS

## Noise, Animal Bioacoustics, and ASA Committee on Standards: Wind Turbine Noise

Nancy S. Timmerman, Cochair
*Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118*

Paul D. Schomer, Cochair
*Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821*

### *Invited Papers*

**8:30**

**3aNS1. A statistical analysis of wind turbine A-weighted sound levels.** Paul D. Schomer (Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com), George Hessler, and David Hessler (Hessler Associates Inc., Haymarket, VA)

Hessler and Hessler collected a two week database of wind turbine 10-min L-90 A-weighted levels in three orthogonal directions from an essentially solitary wind turbine (unit located at end of north-south row). These data have been analyzed statistically and this analysis shows that the wind turbine emissions during the day and at night form a clearly normal distribution with a mean level of 32 dB and standard deviation of 2.4 dB for daytime, 7 Am to 10 PM and a mean level of 36 dB and standard deviation of 2 dB, for nighttime, 11 PM to 5 AM. The nighttime hours were selected as the louder hours of the night when, presumably, an inversion was present. The statistical plots clearly show the data collected for wind turbine non-operation, the transition region between non-operation and full operation, the region of full turbine power, and the data that represent discreet noisier events. This result from a comprehensive, single survey suggests that it is necessary to analyze the noisier nighttime hours separately from daytime or the entire 24-h day, if one is to correctly predict or measure the noise during these critical nighttime hours.

**8:50**

**3aNS2. A critical analysis of the "Wind Turbine Health Impact Study: Report of Independent Expert Panel".** Paul D. Schomer and Pranav K. Pamidighantam (Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

The "Wind Turbine Health Impact Study: Report of Independent Expert Panel" study, herein the "Massachusetts study," says: "The Massachusetts Department of Environmental Protection (MassDEP) in collaboration with the Massachusetts Department of Public Health (MDPH) convened a panel of independent experts to identify any documented or potential health impacts of risks that may be associated with exposure to wind turbines, and, specifically, to facilitate discussion of wind turbines and public health based on scientific findings." It continues to say: "The scope of the Panel's effort was focused on health impacts of wind turbines *per se*." The Massachusetts study treats health affects broadly in accordance with WHO and includes direct health effects, annoyance, and sleep disruption. In many ways, the Massachusetts study is a critique of the literature relating to wind farm acoustic emissions and health effects. This paper is a critique of the critics. In particular, this critique examines some of the physical acoustic findings and some of the social survey findings. The Massachusetts study employed very strict standards to what they deemed to be quality, acceptable studies, and it is only fair that they be judged by their own criteria. It is the judgment of this reviewer that they failed.

**9:10**

**3aNS3. Measuring wind turbine infrasound in the presence of wind.** Richard Carman and Michael Amato (Wilson, Ihrig & Associates, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, rcarman@wiai.com)

A windscreen enables noise measurements to be made as accurately as possible under typical field conditions. Measuring wind turbine noise presents a greater level of difficulty than normal. For there to be wind turbine noise, there needs to be wind for the turbines to operate. Greater wind turbine noise is generally associated with higher wind speeds. Multiple windscreens have been found to significantly reduce artificial wind noise in the range of hearing of 20 to 20 kHz, but cannot reduce the very low frequency pressure fluctuations associated with the movement of air during gusts of wind or in the case of interior measurements fluctuations due to wind pressurizing the building. Wind turbines have been demonstrated by others to produce infrasound in the range of 0.5 to 10 Hz. Measuring infrasound in the presence of local wind is a challenge, since finding a less windy time is not an option. Area-wide measurements of wind turbine noise were conducted at two wind turbine facilities. In analyzing the recorded data, a cross-spectral method was used to reduce the transient effects of local wind in the infrasound range. The technique for doing this is presented and its effect on the data is discussed.

9:30

**3aNS4. Area-wide infrasound measurements for two wind turbine facilities.** Richard Carman and Michael Amato (Wilson, Ihrig & Associates, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, rcarman@wiai.com)

Area-wide measurements of low frequency wind turbine noise were conducted in residential areas adjacent to two different wind turbine facilities in Southern California. The residential measurement location distances ranged from 615 m to 9 km from wind turbines. Additional measurements were also conducted at distances as close as 125 m from the wind turbines. To obtain the residential measurement data, simultaneous digital recordings were made inside and outside residences using microphones designed to achieve a linear response down to 0.07 Hz. The outdoor measurements were conducted with a ground board and two windscreens. The recorded data at residences were analyzed using a cross-spectral technique to minimize the effects of wind acting on the microphone. The data clearly show the presence of infrasound at the blade passage frequency of the wind turbines as well as at the associated harmonics. The primary range of interest is frequencies between 0 and 10 Hz. The residential data in some instances indicate higher levels of infrasound indoors compared to outdoors, indicating a potential amplification of very low frequency sound energy by the residential structure. Representative infrasound data for both facilities are presented and discussed.

9:50–10:00 Break

## *Contributed Papers*

10:00

**3aNS5. Acoustic interaction as a primary cause of infrasonic spinning mode generation and propagation from wind turbines.** Kevin A. Dooley (Kevin Allan Dooley, Inc., 55-1817 Harbour Square, Toronto, ON M5J 2L1, Canada, kadooleyinc@rogers.com) and Andy Metelka (Sound and Vibrations Solutions, Inc., Acton, ON, Canada)

Relatively balanced load and velocity related pressure waves from the rearward facing surface of each rotor blade, are at a frequency of 1 cycle per revolution of the turbine and are phase shifted by 120 degrees from each other. The superposition of these infrasonic waves destructively interfere. This action results in a non-propagating rotor locked mode; however, the shielding (reflecting) effect of the tower as each blade passes, interrupts the balanced destructive interference for a small portion of rotor angle three times per revolution. The momentary un-balance between the destructive interfering waves results in the generation of Tyler-Sofrin spinning mode series, which propagate into the far field. The spinning mode radiation angles, coupled with the low decay rate of infrasound, result in higher far field sound pressure levels than would be predicted for a point source. An analysis approach partially derived from Tyler-Sofrin (1962) is presented. Field microphone data including phase measurements identifying spinning modes are also presented.

10:15

**3aNS6. Significant infrasound levels a previously unrecognized contaminant in landmark motion sickness studies.** Kevin A. Dooley (Kevin Allen Dooley, Inc., 55-1817 Harbour Square, Toronto, ON M5J 2L1, Canada, kadooleyinc@rogers.com)

Airborne Infrasound at any given point can be accurately described as fluctuations or cyclic changes in the local barometric pressure. Variations in a motion sickness test subject's elevation result in fluctuations in the surrounding barometric pressure by similar degrees to that experienced on a ship in high seas. Cyclic variation in the lateral or linear velocity of a subject in a vehicle or platform in atmospheric air may also be subject to infrasonic pressure fluctuations due to the Bernoulli principle and possibly vortex shedding effects. Calculations presented demonstrate that in at least one landmark study (McCauley *et al*., 1976) test subjects were exposed to infrasonic sound pressure levels in excess of 105 db at discrete frequencies between 0.063 and 0.7 Hz. The infrasonic sound pressure level necessarily present in cyclic motion in free atmospheric air does not appear to have been accounted for as a nausea influencing factor in the McCauley *et al*. (1976) motion sickness studies.

10:30

**3aNS7. Narrowband low frequency pressure and vibration inside homes in the proximity to wind farms.** Andy Metelka (Sound and Vib. Solutions Canada Inc., 13652 4th Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

Narrowband fast Fourier transform measurements made outside and inside homes indicate that unlike audible tones, low frequency pressure waves penetrate homes virtually un-attenuated. Simultaneous multi-channel linear weighted pressure measurements were made in various locations as well as seismic floor velocities using various dynamic signal analyzers and sensors. Several areas of concern would include harmonics of blade pass frequencies and also modulation at 20, 30, and 40 Hz (audible). Traditional acoustic models do not predict or measure low frequencies as raw unweighted pressure. Measurements to be presented indicate pressure levels much higher than the audible tone pressure levels. Similar signatures were measured as seismic ground vibration in the basements of homes at relatively low levels. In order to minimize the effects in the homes and to locate wind turbines properly, it may be important to establish measurement standards for low frequencies before locating wind turbine developments.

## Session 3aPAa

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

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

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

### *Invited Papers*

**8:30**

**3aPAa1. On the crest factor of noise from supersonic jets.** Kent L. Gee, Tracianne B. Neilsen (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

An important consideration in characterizing noise from heated, supersonic jets is the crest factor (CF). The large CF in high-speed jet noise is the result of a positively skewed probability density function for the waveform, which translates into infrequently occurring, large-amplitude positive peak pressures. Sufficient system headroom is required in the data acquisition system to provide an accurate representation of these peak pressures and thus avoid clipping or microphone saturation/distortion. But the question remains as to the importance of capturing the single largest pressure out of potentially millions of waveform samples or if a percentile-based CF is adequate. Measurements near a static tactical aircraft reveal CF increases with engine power, with the maximum CF directed upstream of the overall sound pressure level, and a maximum CF of 20 dB at full afterburner. Second, clipping of measured waveforms at different thresholds reveals that a CF definition based on the 99.99 percentile is sufficient to represent overall and band pressure levels to within 0.1 dB and waveform and time-derivative skewnesses to within ~1%. If an estimate of the time-derivative kurtosis is needed within 1% accuracy, then the 99.999 percentile CF is required for headroom estimates.

**8:50**

**3aPAa2. Tactical aircraft noise reduction using fluidic nozzle inserts.** Philip Morris, Dennis McLaughlin, Michael Lurie, and Alex Karns (Aerosp. Eng., Penn State Univ., 233C Hammond Bldg., University Park, PA 16802, pjm@psu.edu)

The noise levels generated by tactical aircraft pose health hazards to personnel working in the vicinity of the aircraft (such as on an aircraft carrier deck) and are annoying to communities close to airbases. The engine exhausts are hot and supersonic and generally operate in an off-design condition, where the nozzle exit and ambient pressures are unequal. This results in shock cells in the jet plume. The interaction between the jet turbulence and the shock cells generates broadband shock-associated noise. The dominant noise radiation is in the downstream direction and is associated with the supersonic convection of turbulence in the jet. This paper describes the development of a technology to reduce both noise sources and involves the controlled injection of air into the diverging section of the nozzle to generate flow corrugations. This enables the jet to operate closer to its design condition and also breaks up the large scale turbulent structures that are responsible for the dominant noise radiation. Both flow and acoustic measurements are described. In addition, steady RANS computations provide information on the flow upstream of the nozzle exit and the effect of injector operating conditions on the flow field. Estimates of nozzle performance are also described.

**9:10**

**3aPAa3. Aeroacoustics of volcanic jets: An overview.** Robin S. Matoza (Scripps Inst. of Oceanogr., Univ. of California, San Diego, IGPP 0225, La Jolla, CA 92093-0225, rmatoza@ucsd.edu), David Fee (Geophysical Inst., Univ. of Alaska Fairbanks, Fairbanks, AK), Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Darcy E. Ogden (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Explosive volcanic eruptions can inject large volumes of ash into heavily traveled air corridors; they pose a significant societal and economic hazard. They also generate large amplitude atmospheric infrasound waves (~0.01-20 Hz), which can be recorded at thousands of kilometers from the eruption and can provide detailed information on the timing, duration, and relative vigor of the volcanic explosions. In order to provide more detail about the eruption process based on acoustic signals, a quantitative model for the acoustic source process within the volcanic eruption column is needed. Volcanic eruption columns are modeled by a momentum-driven jet flow, transitioning with altitude into a thermally buoyant plume. Infrasound recordings from such activity resemble the large-scale turbulence similarity spectrum, indicating that large-scale volcanic jet flows generate an infrasonic form of jet noise. However, volcanic jet noise deviates from pure-air laboratory jet noise because of complexities such as multiphase flow (especially loading with ash particles); nozzle/crater geometry and roughness; buoyancy effects; and high temperature and density effects. We propose a new framework for understanding acoustic sources at volcanoes based on aeroacoustics research, which is being developed through multi-disciplinary integration of field, numerical, and laboratory studies.

9:30

**3aPAa4. High Skewness Infrasound from the eruption of Nabro Volcano, Eritrea: Comparison with supersonic jet and rocket engine data.** David Fee (Geophysical Inst., Univ. of Alaska Fairbanks, 903 Koyukuk Dr, Fairbanks, AK 99775, dfee@gi.alaska.edu), Robin S. Matoza (Inst. of Geophys. and Planetary Phys., Scripps Inst. of Oceanogr., La Jolla, CA), Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Darcy E. Ogden (Inst. of Geophys. and Planetary Phys., Scripps Inst. of Oceanogr., La Jolla, CA)

An understanding of volcanic jets is critical to determining volcanic eruption column dynamics and mitigating volcanic hazards. However, volcanic jets are inherently difficult to observe directly due to their violence, opacity, and complex multi-phase and multi-component flow features. Recent work has shown similarities between the sound produced from explosive volcanic jets and man-made jet engines and rockets. We show that infrasound generated by the 2011 eruption of Nabro Volcano, Eritrea has high waveform skewness and similar waveform statistics to sound produced by supersonic jet engines and rockets. The infrasound from Nabro reported here strongly indicates that infrasound from some volcanic eruptions is produced in similar ways to man-made jet noise from heated, supersonic jet engines and rockets. Noise sources and flow dynamics of jet engines and rockets are better characterized and understood than volcanic jets, suggesting volcanologists could utilize the modeling and physical understandings of man-made jets.

9:50

**3aPAa5. Effective Gol'dberg number for diverging waves.** Mark F. Hamilton (Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Interest in characterizing nonlinearity in jet noise has motivated consideration of an effective Gol'dberg number for diverging waves [Baars and Tinney, Bull. Am. Phys. Soc. **57**, 17 (2012)]. Fenlon [J. Acoust. Soc. Am. **50**, 1299 (1971)] developed expressions for the minimum value of $\Gamma$, the Gol'dberg number as defined for plane waves, for which shock formation occurs in diverging spherical and cylindrical waves. The conditions were deduced from a generalized Khokhlov solution and depend on the ratio $x_{sh}/r_0$, where $r_0$ is source radius, and $x_{sh}$ the plane-wave shock formation distance for $\Gamma=\infty$. Alternatively, by taking the ratio of the nonlinear and thermoviscous terms in Fenlon's Eq. (2), it is proposed here that effective Gol'dberg numbers may be identified for spherical and cylindrical waves: $\Lambda=\Gamma\exp(-\pi x_{sh}/2r_0)$ and $\Lambda=\Gamma/(1+\pi x_{sh}/4r_0)$, respectively. For a given value of $\Lambda$, the diverging waves achieve approximately the same degree of nonlinear distortion as a plane wave for which the value of $\Gamma$ is the same. Conversely, to achieve the same degree of nonlinear distortion as a plane wave with a given value of $\Gamma$, the value of $\Gamma$ for, e.g., a spherical wave must be larger by a factor of $\exp(\pi x_{sh}/2r_0)$. Extensions to other spreading laws are presented.

**10:10–10:30 Break**

## *Contributed Papers*



10:30

**3aPAa6. Nonlinear sound propagation associated with a high mass flow cold jet.** Andrew Marshall and Neal Evans (Southwest Res. Inst., 6220 Culebra Rd., San Antonio, TX 78238-5166, andrew.marshall@swri.org)

It is well-known that aircraft and rocket engines produce high amplitude broadband noise, but such noise can also be generated by high-pressure gas venting from piping systems. Compared to rocket and jet engines, however, the gas exiting these systems can be very low in temperature. During a recent full-scale blow-down test at Southwest Research Institute, noise measurements of a cold jet were obtained. High pressure gas was forced through a valve, pipe, and nozzle system to simulate a natural gas blow-down event in order to measure stresses at welded connections. Nitrogen gas flowed vertically through a 50 cm nozzle with an average mass flow rate of 27.7 kg/s. Noise measurements were made perpendicular to the jet direction at two ranges (18.3 and 157 m). Peak amplitudes of 155 and 138 dB were obtained at the near and far range, respectively. A comparison between this data and rocket engine measurements from the literature will be discussed, including indicators of nonlinear propagation.

10:45

**3aPAa7. Preliminary phased-array characterization of near-field military jet aircraft noise.** Blaine M. Harker, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., 657 E 420 N, Provo, UT 84606, blaineharker@byu.net), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Ashville, NC)

Major developments over the past decade in aeroacoustic beamforming techniques provide more accurate estimates of jet noise source phenomena. In a recent experiment, near and mid-field measurements of an F-22A using linear and planar microphone arrays were taken at various engine conditions about the jet plume. To locate and provide accurate amplitude levels of jet noise sources, conventional beamforming techniques are used with various array shading methods. Equivalent source reconstructions are shown for different engine conditions, observation angles, and frequencies to explore the source region. In addition, different datasets from spatially separated arrays are combined for improved source reconstructions and to account for spatially dependent spectral content. These results are preliminary to further techniques—such as deconvolution methods—to better understand noise source mechanisms within the jet plume. [Work supported by ONR.]

11:00

**3aPAa8. Aeroacoustic source measurement methods for characterizing the sound generated by ducted flow devices with higher-order modes.** Timothy J. Newman, Anurag Agarwal, Ann P. Dowling (Dept. of Eng., Univ. of Cambridge, Trumpington St., Cambridge CB2 1PZ, United Kingdom, tjn25@cam.ac.uk), and Ludovic Desvard (AeroAcoust. Res. Team, Dyson Ltd., Malmesbury, United Kingdom)

The International Organization for Standardization (ISO) method 5136 is widely used in industry and academia to determine the sound power radiated into a duct by fans and other flow devices. The method involves placing the device at the center of a long cylindrical duct with anechoic terminations at each end to eliminate reflections. A single off-axis microphone is used on the inlet and outlet sides that can theoretically capture the plane-wave mode amplitudes but this does not provide enough information to fully account for higher-order modes. In this study, the "two-port" source model is formulated to include higher-order modes and applied for the first three modes. This requires six independent surface pressure measurements on each side or "port." The resulting experimental set-up is much shorter than the ISO rig and does not require anechoic terminations. An array of six external loudspeaker sources is used to characterize the passive part of the two-port model and the set-up provides a framework to account for transmission of higher-order modes through a fan. The relative importance of the higher-order modes has been considered and their effect on inaccuracies when using the ISO method to find source sound power has been analyzed.

11:15

**3aPAa9. Ballistic shock wave localization estimation of shooter position and velocity using difference of time of arrival (DTOA) algorithm in orthogonally arranged discrete acoustic arrays: Part II.** Murray S. Korman (Dept. of Phys., United Stated Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu) and Antal A. Sarkady (Dept. of Elec. and Comput. Eng., U.S. Naval Acad., Annapolis, MD)

A mathematical algorithm was developed to estimate the parameter space involving both the displacement of the shooter with respect to the array and the velocity of the projectile using the difference in time of arrival, DTOA, of the ballistic shock wave cone at each position of an N-element array. [J. Acoust. Soc. Am. **133**, p. 3506, May 2013.] The array geometry involves orthogonally arranged discrete point-like line arrays using wide-band microphone or piezo-electric elements. The algorithm utilizes a nonlinear least squares parameter fit by summing the squares of [DTOA (experimental)—DTOA (theoretical parameters)] values where the DTOA (theoretical) equation involves a lengthy Taylor series expansion of the exact "difference in time of arrival" theoretical equation. Earlier results, in the absence of noise ($N=7$) showed that the model has good versatility in estimating displacement (location) and velocity in simulated computer trials. Here, near-field noise is simulated by a rotor blade resulting in uncertainty in the arrival time of the shockwave at each sensor, leading to uncertainty in the estimation of the fit parameters. For example, simulated scatter plots of the azimuthal angle vs the elevation angle parameters for different projectile miss distances become useful in computing parameter uncertainty for different signal-to-noise ratios.

11:30

**3aPAa10. Nonlinear acoustics of combustion instability in solid-propellant rocket motors.** Hunki Lee, Taeyoung Park, Won-Suk Ohm (Yonsei Univ., Seoul, South Korea), and Dohyung Lee (Agency for Defense Development, Daejeon, South Korea)

Combustion instability, a large oscillation of pressure in a combustion chamber, is known to be a major source of rocket failure. A common approach to analyzing combustion instability is to regard it as an acoustical phenomenon in an enclosure, driven by the combustion process occurring in a thin region near the grain (solid propellant). Because of the large pressure excursion associated with combustion instability, it exhibits many salient features of nonlinear wave process such as waveform distortion, shock formation, and even chaotic behaviors. In this paper, a comprehensive analytic model for combustion instability of a solid rocket motor is presented. Our focus is on the way in which nonlinearity manifests itself under complex grain geometry, where the acoustic modes can be either harmonically or anharmonically related. Predictions from the model are compared with the static test data for a few representative rockets.

WEDNESDAY MORNING, 4 DECEMBER 2013 CONTINENTAL 2/3, 8:00 A.M. TO 9:55 A.M.

## Session 3aPAb

## Physical Acoustics: Phonons and Lattice Dynamics

Veerle M. Keppens, Cochair
*Univ. of Tennessee*

Josh R. Gladden, Cochair
*Phys. & NCPA, Univ. of Mississippi, 108 Lewis Hall, University, MS 38677*

**Chair's Introduction—8:00**

### *Invited Papers*

8:05

**3aPAb1. Glass-like phonon scattering from spontaneous nanostructures in silver-antimony-tellurium.** Jie Ma (Quantum Condensed Matter Div., Oak Ridge National Lab., PO BOX 2008 M.S. 6430, Oak Ridge, TN 37831, jea@ornl.gov), Olivier Delaire, Andrew May (Mater. Sci. and Technol. Div., Oak Ridge National Lab., Oak Ridge, TN), Chris Carlton (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), Michael McGuire (Mater. Sci. and Technol. Div., Oak Ridge National Lab., Oak Ridge, TN), Lindsay VanBebber (Dept. of Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN), Douglas Abernathy, Georg Ehlers, Tao Hong (Quantum Condensed Matter Div., Oak Ridge National Lab., Oak Ridge, TN), Ashfia Huq (Chemical and Eng. Mater. Div., Oak Ridge National Lab., Oak Ridge, TN), Wei Tian (Quantum Condensed Matter Div., Oak Ridge National Lab., Oak Ridge, TN), Veerle Keppens (Dept. of Mater. Sci. and Eng., Univ. of Tennessee, Knoxville, TN), Shao-Horn Yang (Dept. of Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Brian Sales (Mater. Sci. and Technol. Div., Oak Ridge National Lab., Oak Ridge, TN)

Materials with very low thermal conductivity are of great interest for both thermoelectric and optical phase-change applications. Synthetic nanostructuring is most promising for suppressing thermal conductivity arising from scattering phonons, but challenges remain in producing bulk samples. In crystalline $AgSbTe_2$, we show that a spontaneously forming nanostructure leads to a suppression of thermal conductivity to a glass-like level. Our mapping of the phonon mean-free-paths provides a novel bottom-up microscopic account of thermal conductivity and also reveals intrinsic anisotropies associated with the nanostructure. Ground-state degeneracy in $AgSbTe_2$ leads to the natural formation of nanoscale domains with different orderings on the cation sublattice, and correlated atomic displacements, which efficiently scatter phonons. This mechanism is general and suggests a new avenue for the nanoscale engineering of materials to achieve low thermal conductivities for efficient thermoelectric converters and phase-change memory devices.

**8:25**

**3aPAb2. Inelastic scattering and resonance ultrasound spectroscopy for functional materials studies.** Raphael P. Hermann (Juelich Ctr. for Neutron Sci. JCNS and Peter Gruenberg Institut PGI, JARA-FIT, Forschungszentrum Juelich GmbH, Leo Brand Str. 1, Juelich 52425, Germany, r.hermann@fz-juelich.de)

The combined use of inelastic scattering, resonant ultrasound spectroscopy, and other macroscopic thermodynamics characterization techniques will be presented. In particular, inelastic neutron scattering and nuclear inelastic scattering (NIS) by Mössbauer resonant nuclei are two techniques that probe acoustic phonons and provide a microscopic counterpoint to direct speed of sound measurements. Results from new developments for the Sb and Te element specific NIS with sub-meV resolution will be presented for materials with thermoelectric or phase change properties. Specifically the lattice softening in the YbFe4Sb12 skutterudite [Moechel *et al*., Phys. Rev. B **84**, 184306 (2011)] and the systematic softening observed in nanostructured thermoelectric materials [Claudio *et al*., J. Mater. Sci. **48**, 2836 (2013)] with respect to their bulk counterpart will be discussed, as well as lattice softening in magnetocaloric MnFe4Si3. [The European Synchrotron Radiation Facility is acknowledged for provision of the synchrotron radiation facility at beamlines ID18 and ID22N; the Institut Laue Langevin for beamtime at IN6; the Deutsche Forschungsgemeinschaft for funding SPP-1386 "Nanostructured Thermoelectrics" and SFB-907 "Nanoswitches"; the BMBF for NanoKoCh 03X3540; and the Helmholtz Gemeinschaft Deutscher Forschungszentren for VH-NG-407 and HRJRG-402.]

**8:45**

**3aPAb3. A long-sought phase transition in superconducting cuprates observed via resonant ultrasound spectroscopy.** Albert Migliori, Arkady Shekhter, Brad J. Ramshaw, and Ross D. McDonald (NSEC-NHMFL, Los Alamos National Lab., M.S. E536, Los Alamos, NM 87545, migliori@lanl.gov)

Among the biggest mysteries of high-temperature superconductors is the so-called pseudogap—somewhat similar to the gap in the electronic density of states found in the superconducting phase, but occurring at a different temperature. The pseudogap may represent either the gradual onset of a precursor to superconductivity or an entirely new phase, characterized by the gain or loss of some hidden order. Several experiments in recent years have favored the latter, but the smoking gun, the thermodynamic signature of a pseudogap phase transition, had not been observed. Using resonant ultrasound spectroscopy, we measured the temperature-dependent elastic stiffness of two cuprate superconducting crystals, one underdoped and one overdoped and found a break in slope at a doping-dependent temperature T*. For the underdoped cuprate, T* coincides with the onset of the pseudogap and with earlier neutron-scattering measurements of the appearance of magnetic order (blue squares). Crucially, for the overdoped cuprate, $T^* < T_c$ so that extrapolating to higher doping where $T^* = 0$ will yield a quantum critical point, which may be key to understanding the mechanism of high-temperature superconductivity (Nature **498**, 75 (2013).



**9:05**

**3aPAb4. Hypersound in simple one-dimensional device structures.** G. Todd Andrews (Phys. and Physical Oceanogr., Memorial Univ., Prince Philip Dr., St. John's, NF A1B 3X7, Canada, tandrews@mun.ca)

Brillouin spectroscopy, an inelastic laser light scattering technique capable of probing long wavelength acoustic phonons in a variety of material systems, was used to study hypersound in simple one-dimensional mesoporous silicon-based device structures formed using electrochemical etching methods. Brillouin spectra of porous silicon superlattices with binary periodicity on the order of the hypersound wavelength reveal zone folding, band gaps, and localized modes, indicating that these structures behave as hypersonic phononic crystals. Superlattices with smaller modulation wavelengths act as effective elastic media. New results on the behavior of hypersound in stacked superlattices and those with deliberately introduced defects will also be presented. Collectively, these studies have led to an improved fundamental understanding of classical wave behavior and interaction in low-dimensional systems and open up exciting opportunities for phonon engineering in a silicon-based platform.

## *Contributed Papers*

**9:25**

**3aPAb5. High temperature elastic constants of rare-earth doped $Sr_{0.9}X_{0.1}TiO_{3-\delta}$ (X = Pr, Y).** Josh R. Gladden, Sumudu P. Tennakoon (Phys. & NCPA, Univ. of MS, NCPA, 1 Coliseum Dr., University, MS 38677, sptennak@go.olemiss.edu), Rasheed Adebisi (SOAIR, LLC, University, MS), Qin Zhang (Phys. & NCPA, Univ. of MS, University, MS), A. M. Dehkordi (Mater. Sci. and Eng., Clemson Univ., Clemson, SC), S. Bhattacharya, T. M. Tritt (Phys. and Astronomy, Clemson Univ., Clemson, SC), and H. N. Alshareef (Mater. Sci. and Eng., King Abdullah Univ. of Sci. and Technol., Thuwal, Saudi Arabia)

Temperature dependence of the elastic constants of polycrystalline rare-earth doped strontium titanate (STO) [ $Sr_{0.9}X_{0.1}TiO_{3-\delta}$ (X = Pr, Y)] was investigated in the temperature range of 300 K—750 K using resonant ultrasound spectroscopy. Elastic constants of undoped STO decrease linearly indicating typical softening with increased temperature. Yttrium (Y) doped STO also exhibits a monotonic softening, however, with a pronounced curvature in this high temperature regime. Trends of elastic constants of the praseodymium (Pr) doped STO show a non-monotonic stiffing from room temperature to 475 K, followed by a gradual softening. Changes in attenuation were quantified by the inverse quality factor (1/Q) averaged over measured resonances. Undoped STO showed a monotonic gradual increase of attenuation with increasing temperature while yttrium doped STO showed little variation. In contrast, attenuation of Pr doped STO exhibited a peak around 425 K. These results will be compared to thermal conductivity measurements in the same temperature range and phonon scattering mechanisms will be discussed.

9:40

**3aPAb6. Capacitive micromachined ultrasonic transducers as tunable phononic crystals.** Shane Lani (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 1454 Catherine St., Decatur, GA 30030, swl5059@gmail.com), M. Wasequr Rashid (School of Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), Karim G. Sabra, and F. Levent Degertekin (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Capacitive micromachined ultrasonic transducer (CMUT) arrays are made up of microscale (10-100μm wide) membranes with embedded electrodes for electrostatic excitation and detection of acoustic waves. While typically used for far-field imaging, CMUT arrays also support dispersive evanescent surface waves. These surface waves derive their dispersive properties not only from the periodic structure of the array, but also from the membrane resonance. One advantage of CMUTs as a metamaterial is that the dispersive qualities of the array can be tuned by changing the applied bias voltage to the membranes, which in effect changes the membrane stiffness. A second advantage is that the CMUT array elements can be used as receivers to record the acoustic waves with high spatial resolution, which make laser displacement measurement based characterization unnecessary. These properties allow the possibility of CMUTs to exploit these slowly propagating evanescent waves as a means for creating subwavelength resolution fields for high-resolution ultrasound imaging and sensing in the near field by appropriately tuning the physical characteristics of individual membranes. The dispersive behavior of these evanescent surface waves propagating along a CMUT array was quantified using a computationally efficient, boundary element method based model and validated with both finite element analysis and experimental data obtained from a 1 x 16 CMUT array with a membrane resonance tunable between 5 and 6.5 MHz.

WEDNESDAY MORNING, 4 DECEMBER 2013 — POWELL, 8:00 A.M. TO 9:40 A.M.

# Session 3aSAa

## Structural Acoustics and Vibration, Noise, and ASA Committee on Standards: Groundborne/Structureborne Noise and Vibration from Transportation

James T. Nelson, Chair
*Wilson, Ihrig & Associates, 6001 Shellmound, Emeryville, CA 94608*

**Chair's Introduction—8:00**

### *Invited Papers*

**8:05**

**3aSAa1. Noise reduction performance of wheel vibration absorbers.** James T. Nelson (Wilson, Ihrig & Associates, 6001 Shellmound, Emeryville, CA 94608, jnelson@wiai.com)

The noise reduction effectiveness of wheel vibration absorbers was evaluated at the Bay Area Rapid Transit System (BART). The vibration absorbers were fitted to the steel tires of aluminum-centered wheels. Wayside noise was recorded at tangent and curved track with resilient direct fixation fasteners, and under-car noise data were recorded throughout the system. Data were collected with and without wheel vibration absorbers installed. The wheel vibration absorbers had little effect on wayside and under-car noise at audio frequencies, though a minor reduction of noise at about 500 Hz was observed, suggesting a change in wheel dynamics. Third octave band noise reductions are summarized for on-board and wayside measurements.

**8:25**

**3aSAa2. Noise reduction performance of wheel and rail vibration absorbers.** Thom Bergen (Wilson, Ihrig & Associates, Inc., 15719 165th Pl. NE, Woodinville, WA 98072, tbergen@wiai.com) and James T. Nelson (Wilson, Ihrig & Associates, Inc., Emeryville, CA)

The rolling noise reduction effectiveness of wheel and rail vibration absorbers was evaluated at Trimet in Portland, Oregon. Tests included tangent ballast and concrete tie track and curved track slab track with bi-block concrete ties. Under-car noise data were recorded throughout the system with and without wheel vibration absorbers. Results of the rolling noise tests are summarized for various combinations of treated and untreated rails and wheels. The wheel and especially the rail vibration absorbers significantly reduced rail vibration at audio frequencies, but had little effect on wayside rolling noise. Wayside noise at tangent track was slightly higher with rail vibration absorbers than without. However, the "singing rail" noise was eliminated entirely. The "singing rail" vertical vibration transmission spectrum had a pass-band characteristic as expected for periodic supports. The vibration data indicate that above 250 Hz the wheel was the dominant source of noise relative to the rail. A reduction of stick-slip noise was observed over most curves, though results were inconsistent.

8:45

**3aSAa3. Groundborne noise produced by rail transit tunnel construction.** Thom Bergen (Wilson, Ihrig & Associates, Inc., 15719 165th Pl. NE, Woodinville, WA 98072, tbergen@wiai.com), James T. Nelson, and Derek L. Watry (Wilson, Ihrig & Associates, Inc., Emeryville, CA)

Groundborne noise and vibration produced by rail transit tunnel construction activities was studied recently in Seattle, Washington. Recent and ongoing tunneling is accomplished with tunnel boring machines (TBMs) and associated supply trains. Vibration produced by construction activities in the tunnels 30 to 40 m below the surface propagated efficiently to through the soils in the Seattle area. The local geology consists largely of "overconsolidated glacial till" that is very stiff. The study revealed that the supply trains traveling over rail joints was the dominant source of ground vibration, and the re-radiated groundborne noise in numerous homes above the tunnels was audible. The noise was effectively mitigated by patching and smoothing out the rail joints, and supporting the ties on natural rubber pads.

9:05

**3aSAa4. Line source response from geotechnical data.** Gary Glickman (Wilson, Ihrig & Associates, 65 Broadway Ste. 401, New York, NY 10006, gglickman@wiai.com)

Predicting groundborne noise and vibration environmental impacts associated with rail transit projects involves determination of the line source response (LSR) to characterize ground vibration propagation characteristics. The presentation discusses the use of geotechnical data to develop model LSRs. GIS is then used to apply model LSR data on a larger scale and perform a detailed analysis using Federal Transit Administration (FTA) methodology, which can be refined with field testing at a later stage. Conceptual track vibration mitigation measures are discussed for controlling groundborne noise and vibration to nearby receptors.

## *Contributed Paper*

9:25

**3aSAa5. Measurement of dynamic viscoelastic properties of flexible polyurethane foam under compression for application to seat vibration analysis.** Deokman Kim, Won-Sok Yoon (School of Mech. Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul, 133-791, Korea, Seoul ASI | KR | KS013, South Korea, deokman@hanyang.ac.kr), Hyun-kyu Park (Res. & Development Div., HYUNDAI·KIA MOTORS, Seoul, South Korea), Hak-Sung Kim, and Junhong Park (School of Mech. Eng., Hanyang Univ., Seoul, South Korea)

Supporting stiffness of the seat is one of the important components affecting dynamic characteristics recognized by a passenger. To analyze dynamic characteristic of seat for vehicles operating on various road conditions, the seat should be understood together with the oscillation due to road irregularity. In this study, the viscoelastic properties of flexible polyurethane foam under compression was measured and used in estimating the dynamic characteristic of seat analyzed as a simplified geometry. The beam transfer function method is used to obtain the dynamic properties of the foam under compression. The viscoelastic properties were obtained to the maximum compression level of 70%. The simple seat model was composed rigid base, edge blocks, elastic supports, and flexible polyurethane foam, and the method is used in the same way to obtain the dynamic support properties. The equivalent support stiffness was estimated on various locations on the seat, and the effects of each component such as the compression level, the foam type, and the stretch of elastic support were analyzed.

# Session 3aSAb

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

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

Wolfgang Grill, Cochair
*ASI Analog Speed Instruments GmbH, Burgweg 8, Koenigstein im Taunus 61462, Germany*

**Chair's Introduction—10:00**

### *Invited Papers*

**10:05**

**3aSAb1. Analysis of acoustic harmonic generation in a solid with multiple nonlinear interfaces.** Shiro Biwa and Yosuke Ishii (Dept. of Aeronautics and Astronautics, Kyoto Univ., C-Cluster III, Katsura, Nishikyo-ku, Kyoto 615-8540, Japan, biwa@kuaero.kyoto-u.ac.jp)

Weak bonds and delaminations are typical examples of imperfect interfaces in multilayered structures. Nonlinear acoustic/ultrasonic methods are expected to offer a promising means to monitor these imperfect interfaces, as such interfaces behave nonlinearly when subjected to high-amplitude waves and result in the occurrence of nonlinear frequency components such as higher harmonics. Harmonic generation at a single nonlinear interface has been studied by many investigators from both theoretical and experimental points of view. In this presentation, a theoretical analysis of harmonic generation at multiple nonlinear interfaces is presented within a framework of one-dimensional elastic wave propagation in the frequency domain. The analysis is based on the perturbation expansion of the wave field by assuming the weak nonlinearity. Specifically, the second-harmonic generation is analyzed by first solving the linear transmission of the incident fundamental component, and then the propagation of the second-harmonic components generated at nonlinear interfaces. Some numerical results are demonstrated and compared to the results of time-domain analysis using the finite element method. The present analysis shows that harmonic generation in multilayered solids is remarkably frequency-dependent, as both the fundamental and the harmonic components interact with the layered structure in a complex manner.

**10:25**

**3aSAb2. Monitoring material nonlinearity and attenuation variations in mortar subjected to freezing-thawing cycles.** Jesus N. Eiras (Instituto de Ciencia y Tecnología del Hormigón (ICITECH), Universitat Politècnica de València, Valencia, Spain), Tribikram Kundu (Civil Eng. & Eng. Mech., Univ. of Arizona, 1209 E. 2nd St., Bldg. # 72, Tucson, AZ 85721, tkundu@email.arizona.edu), John S. Popovics (Civil and Environ. Eng., Univ. of Illinois, Urbana, IL), J. Monzó, M. V. Borrachero, and J. Payá (Instituto de Ciencia y Tecnología del Hormigón (ICITECH), Universitat Politècnica de València, Valencia, Spain)

Standard vibration resonance frequency tests have been widely used for prediction of material modulus of elasticity and for monitoring damage in cement-based materials. More recently, dynamic non-classical nonlinear analyses show promise for damage diagnosis through a variety of test methods that are generally called nonlinear elastic wave spectroscopy (NEWS) techniques. In this study, we monitor the nonlinear dynamic behavior and mechanical wave attenuation of mortar subjected to varying numbers of freezing-thawing cycles. The nonlinear analysis is deployed using a new signal processing technique applied to standard resonance frequency test generated data. The proposed technique is demonstrated on damaged and un-damaged mortar bar samples.

**10:45**

**3aSAb3. Nonlinear ultrasonic waves for monitoring thermal stresses in solids.** Claudio Nucera and Francesco Lanza di Scalea (Structural Eng., Univ. of California San Diego, 9500 Gilman Dr., MC 0085, La Jolla, CA 92093, flanza@ucsd.edu)

It is known that nonlinear ultrasonic waves in solids are sensitive to quasi-static stresses. The stress sensitivity of elastic waves is typically associated to finite strains (e.g., theory of acoustoelasticity). In the case of waveguides, classical nonlinear theories for guided waves are still based on the assumption of finite strains. In the case of constrained solids subjected to thermal excursions, however, there are theoretically no finite strains (for perfectly constrained solids) associated with thermal stresses. A new model is therefore needed to justify the existence of wave nonlinearities in this case of stress without strain. This problem is solved on the basis of the interatomic potential of the solid that indicates a "residual" strain energy, due to the prevented thermal expansion, which is at least cubic as a function of strain. Consequently, a nonlinear wave equation can be derived. The solution to this equation leads to a new nonlinear parameter for double harmonic generation that is directly related to the thermal stresses in the structure. This study finds applications in the monitoring of thermal stresses in buckling-prone structures, such as continuously welded railroad tracks and pipelines. Experimental tests conducted on railroad tracks with realistic support will be also presented.

11:05

**3aSAb4. Noncontact fatigue crack detection using nonlinear wave modulation spectroscopy.** Peipei Liu, Hoon Sohn (Dept. of Civil and Environ. Eng., KAIST, Deajeon, South Korea,, hoonsohn@kaist.ac.kr), and Tribikram Kundu (Dept. of Civil Eng. and Eng. Mech., Univ. of Arizona, Tucson, AZ)

Nonlinear wave modulation spectroscopy (NWMS) has been used to evaluate nonlinear acoustic signature of fatigue cracks in materials and thus to get an idea about the degree of material nonlinearity. It is done by generating ultrasonic waves at two different frequencies and measuring their modulation. The choice of two distinct frequencies plays a significant role in NWMS for different structures. In this paper, instead of using signals at two distinct frequencies, only one broadband pulse signal is used as the driving signal, which can be generated by a laser beam. This driving signal generates multi-frequency responses as different resonance frequency modes and/or Lamb wave modes, generated in a plate-like structure. Nonlinear wave modulation occurs among these frequencies when material nonlinearity exists. It increases the sideband energy and the number of peaks in the spectral plots. These two features, namely sideband energy ratio (SER) and sideband peak number (SPN), are extracted from the spectral plots to measure the material nonlinearity caused by fatigue cracks. The noncontact laser system has been built for NWMS measurement by integrating and synchronizing a Q-switched Nd:YAG laser for ultrasonic wave generation and a laser Doppler vibrometer for ultrasonic wave detection. The proposed modified NWMS technique with the noncontact laser system has been successfully used for the identification of metallic plates with fatigue cracks.

11:25

**3aSAb5. "Incubation of damage" state quantification in laminated composites and metallic alloys.** Sourav Banerjee (Dept. of Mech. Eng., Univ. of South Carolina, 300 Main St., Rm. A117, Columbia, SC 29208, banerjes@cec.sc.edu)

In this presentation, a comparatively simple but efficient novel approach is proposed to quantify the "incubation of damage" state using scanning acoustic microscope (SAM). The proposed approach exploits the nonlocal micromorphic field theory to quantify intrinsic (multi-scale) damage state. Defying the conventional route of 'bottom-up' multi-scale modeling methods, a hybrid 'top-down' approach is presented, which is then correlated to ultrasonic signature obtained from composite and metallic alloy specimens. A parameter to quantify the incubation of damage at meso-scale has been identified in this paper. The intrinsic length scale dependent 'parameter called 'damage entropy' closely resembles the material state due to fatigue, extreme environments, operational hazards or spatio-temporal variability, etc. The proposed quantification process involves fusion between micromorphic physics and high frequency ultrasonic. The proposed approach is validated through an experimental study conducted on sequentially fatigued glass-fiber reinforced polymer composites and Aluminum 5xxx aluminum alloy specimens. Specimens were characterized under scanning acoustic microscope (50 and 100 MHz). The imaging data and the sensor signals are characterized to quantify the incubation of damage state by a new parameter called 'damage entropy.'



WEDNESDAY MORNING, 4 DECEMBER 2013 PLAZA B, 8:30 A.M. TO 12:00 NOON

## Session 3aSC

## Speech Communication: Acquiring Speech: Children and Adults

Catherine L. Rogers, Chair

*Communication Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620*

### *Contributed Papers*

8:30

**3aSC1. Younger versus older infants' use of prosody-like boundaries to locate musical phrases.** Kara E. Hawthorne (Linguist, Univ. of AB, 2-40 Assiniboia Hall, Univ. of Alberta, Edmonton, AB T6G 2E7, Canada, kara.hawthorne@gmail.com) and LouAnn Gerken (Linguist, Univ. of Arizona, Tucson, AZ)

In spoken language, particularly of the infant-directed variety, clauses are marked with prosodic cues, such as final lengthening and pitch resets at boundaries (e.g., Soderstrom *et al*., 2008). Though prosody is specific to language, similar acoustic cues mark phrase boundaries in music, and infants are sensitive to the correlation of these cues at musical boundaries (Jusczyk and Krumhansl, 1993). In the present study, we ask whether younger (4-month-old) and older (16-month-old) infants can use prosody-like boundary cues to facilitate recognition of musical phrases. In a musical extension of Soderstrom *et al*. (2005), infants were familiarized with one of two brief melodies, then were tested using the head-turn preference procedure on their ability to recognize a phrase versus phrase-straddling excerpt from the familiarization melody when it is embedded in a new musical passage. Preliminary results suggest that younger infants show a familiarity preference for the test melody containing a phrase from the familiarized melody, while older infants do not discriminate between the test item types. This suggests that young infants use prosody-like cues to group acoustic stimuli in a domain-general way, whereas older infants may not.

**8:45**

**3aSC2. Assessing language acquisition from parent-child interaction: An event-related potential study on perception of audio-visual cues in infancy.** Eeva Klintfors, Lisa Gustavsson, Iris-Corinna Schwarz, Tove Gerholm, and Ulrika Marklund (Linguistics/Phonet., Stockholm Univ., Universitetsv 10 C, Stockholm SE – 106 91, Sweden, eevak@ling.su.se)

This paper promotes a theory-driven model development of parent-child interaction. In our project, we identify, test, and simulate some of the fundamental components of speech, gestures, and social-emotional behaviors and the consequences they might have on child language development. Our theoretical position is part of the connectionist tradition; language acquisition is described to be an emergent consequence of the interplay between the infant and the ambient linguistic environment, including sensory information of all modalities. It is well known that speech comprehension and production are significantly influenced by the presence of co-speech gestures. These gestures may be articulatory in nature or hand/beat co-gestures that keep the rhythm of speech. However, since the extent of this integrated relationship is difficult to determine from behavioral research solely, studies addressing neural mechanisms that underlie cognitive processes and behaviors are of importance. This paper reports an electroencephalography/event-related potential (EEG/ERP) pilot study on children's early perception of congruent versus incongruent audio-visual pairings (e.g., acoustic information matching vs not matching the articulation shown). Ultimately, it is our hope that understanding the integrated speech-gesture relationship may provide insights into how children allocate resources while speaking and help clinicians/teachers to better identify and treat children with developmental disorders.

**9:00**

**3aSC3. Assessing language acquisition from parent-child interaction: An event-related potential study on perception of intonation contours in infancy.** Lisa Gustavsson, Eeva Klintfors, Iris-Corinna Schwarz, Tove Gerholm, and Ulrika Marklund (Linguistics/Phonet., Stockholm Univ., Universitetsv 10 C, Stockholm SE-106 91, Sweden, lisag@ling.su.se)

The aim of this paper is to present our multidisciplinary project to study parent-child interaction. The goal of the project is to identify, test, and simulate components of child and adult speech and gestures and the consequences they might have on child language acquisition. Since typical parent-child interaction is built upon both interlocutors' intention-reading, responsiveness to joint-attention, and imitation of speech/gestures, we make video recordings along with recordings of speech data to grasp the integration of semantic and pragmatic aspects of language acquisition. The understanding of parent-child interaction benefits further from information on brain activation involved in speech processing. As a first step to achieve the project goals, an electroencephalography/event-related potential (EEG/ERP) study exploring children's early perception of intonation contours involved in human interactions was performed. This paper discusses the characteristics of integration of multimodal social-emotional (speech, prosody, faces, posture) signals as part of the dynamics of communication in typically developing children. Possible application fields are social signal processing (SSP; an emerging research domain that aims to provide computers ability to understand human social signals), and improvement of diagnosis of late or atypical language development in pathologies that affect the dynamics of social interaction (such as autism spectrum disorders).

**9:15**

**3aSC4. Stop production in bilingual and second language-learning children.** Sue Ann Lee (Texas Tech Univ Health Sci. Ctr., 3601 4th St., Lubbock, TX 79430, sueann.lee@ttuhsc.edu), Gregory Iverson (Univ. of Wisconsin, Milwaukee, WI), and Jahyung Lee (Ewha Womans Univ., Seoul, South Korea)

This study examined stops produced by 7 year-old Korean-English bilingual (KEB) children and age-equivalent Korean children who had learned English as a second language (L2) in order to investigate how duration of exposure affects the PHONETIC systems of their two languages. A total of 60 children participated (15 per group; monolingual English, monolingual Korean, KEB and L2 children). Word-initial VOT and f0 values in the following vowel were measured in both languages. Comparison of English and Korean stops produced by monolingual children showed that the two English (voiced and voiceless) and three Korean (fortis, lenis, & aspirated) stop types were fully distinguished. Like the monolinguals, KEB children produced English and Korean stops distinctively, indicating that they possess two separate stop systems. But while L2-learning children distinguished English voiced from Korean fortis, and English voiceless from Korean lenis, they produced English voiceless and Korean aspirated stops similarly. Compared to adult Korean L2 learners who did not distinguish English voiced from Korean fortis (Kang and Guion, 2006), the results here suggest that young L2 children express more sophisticated phonetic categories than do adult L2 learners. [Funded by NICHD (RHD061527A).]

**9:30**

**3aSC5. The role of orthographic information in the learning of allophonic variation.** Chung-Lin Yang (Linguist, Indiana Univ., 2100 E Lingelbach LN Apt. 503, Bloomington, IN 47408, cy1@indiana.edu) and Isabelle Darcy (Second Lang. Studies, Indiana Univ., Bloomington, IN)

Exposure to L2 orthography may facilitate learning a novel vocalic (e.g., Escudero *et al.*, 2008) or tonal (Showalter and Hayes-Harb, 2013) L2 contrast. Yet it is unclear whether the benefit of orthographic information applies to the learning of L2 words involving allophonic variants. We investigated whether exposure to L2 orthography can help L2 learners establish a single lexical representation for words containing allophones. We used an invented language, with word-pairs of free variants (test condition) involving the vowel alternation [ɔ]-[u], both of which can be spelled as <o>. In the control condition, vowel alternation [e]-[a] contrasted word meanings. In a word learning experiment, Mandarin and American English speakers were presented with words paired with pictures. In addition, one subgroup of participants saw the spellings when they heard the words, while another did not. Then, in a picture-auditory word matching task, participants who learned that the variants were allophonic were expected to link the two variants to one single picture in the test condition only, not in the control. A facilitative effect of orthography on the learning of free variation was observed for Mandarin speakers. This shows that orthography may help L2 learners establish a single lexical representation for allophonic variants.

**9:45–10:00 General Discussion**

**10:00–10:30 Break**

**10:30**

**3aSC6. Difficulty in the acquisition of Mandarin high level and high falling tones by Cantonese learners.** Xianghua Wu (Dept. of East Asian Lang. and Cultures, Univ. of California,, 3110 Dwinelle,Berkeley, CA 94720-2230, xianghwa.wu@gmail.com) and Kazuya Saito (School of Commerce, Waseda Univ., Shinjuku, Japan)

Native speakers of tone languages commonly have difficulty discriminating tones with the same phonological function (Huang, 2001), such as Cantonese high level and high falling tones; however, acquisition of such tones with distinctive phonological status in a second language (L2) remains unclear. This study tested 34 Cantonese learners before and after training on Mandarin high level, mid-rising, and high falling tones. Perception was evaluated using a forced-choice identification task, and nine native speakers of Mandarin judged productions from repetition and narrative tasks. Despite improvement in post-tests, perception of high level and high falling tones was found to be more difficult than mid-rising tone in both pre- and post-tests. Misidentification patterns also showed more confusion between high level and high falling tones than other tone pairings, but no effect of training was observed. Compared to the perception results, high falling tone was produced more frequently as high level tone, particularly in the narrative task before and after the training. The results suggest that L2 tone acquisition is complicated by the complex phonological correspondence between L1 and L2 tones. [Research supported by Language Learning Research Grant.]

10:45

**3aSC7. Development of vowel spaces from age 21 to age 49 in a group of 8 talkers.** Auburn Lutzross, William Schuerman, Ronald Sprouse, and Susanne Gahl (Linguist, Univ. of California Berkeley, 2435 Grant St., Apt. 2, Berkeley, CA 94703, alutzross@berkeley.edu)

We describe age-related change in speech during young to middle age adulthood using a new resource for phonetic and sociolinguistic analysis. This resource is based on the "Up" series of documentary films, showing a set of 11 individuals filmed at seven year intervals over a period of 42 years. We analyzed 67 sample utterances (minimum duration = 10 s), containing 4493 vowels produced by eight talkers, with the aim of understanding how vowel spaces change in young and middle-age adulthood, prior to physiological changes often observed in elderly talkers. We measured the first and second formants of each vowel token and analyzed several measures of vowel distribution in F1/F2 space: Euclidean distance from the talker's average F1/F2 ("dispersion"), within-category variability (intra-vowel dispersion), and vowel space area. Area of the vowel space was measured using averages of point vowels (/a/, /ae/, /i/, /u/), as well using the convex hull of all vowels. For some individuals, vowel spaces from age 21 to age 49 come to be more compact, in a manner that has previously been observed in elderly speakers. However, we also find considerable individual variability, with no clear age-related trend across speakers.

11:00

**3aSC8. Non-native vowel production accuracy and variability in relation to overall intelligibility.** Svetlin Dimov and Ann Bradlow (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208-4090, svetlindimov2011@u.northwestern.edu)

Previous research suggests that accuracy (i.e., distance to the average location of native productions) has less effect on adaption to non-native speech than category variability [e.g., Wade *et al*., Phonetica **64**, 122-144 (2007)]. Here we investigate the relationship between overall intelligibility of Mandarin-accented English for native English listeners and (a) vowel production accuracy, and (b) vowel production consistency. Intelligibility estimates were based on sentence-in-noise recognition accuracy scores. Vowel accuracy and consistency estimates were based on formant measurements of point vowels (/i/, /u/, /ae/, and /a/) extracted from words in the sentence materials that were presented to listeners for intelligibility testing (8-20 samples/vowel/talker). If listeners have expectations about a vowel category location based on accumulated exemplar storage, then greater accuracy (smaller Euclidean distance to native category mean) should be positively related to intelligibility. If listeners are sensitive to vowel category distributions, then greater consistency (smaller standard deviation of f1 or f2 within categories) should be beneficial to intelligibility. A mixed effects linear model revealed that only consistency was a significant predictor of intelligibility. Accuracy was not a significant predictor. This result suggests that intra-speaker variability is detrimental to L2 intelligibility, regardless of distance to native categories.

11:15

**3aSC9. Individual differences in learning to perceive novel phonetic contrasts: How stable are they across time and paradigms?** Mirjam Broersma (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, P.O. Box 9103, Nijmegen 6500 HD, Netherlands, m.broersma@let.ru.nl), Dan Dediu, and Jiyoun Choi (Max Planck Inst. for PsychoLinguist, Nijmegen, Netherlands)

Previous research has shown that learners differ widely in the success with which they learn to perceive novel phonetic contrasts. Little is known, however, about the stability of such differences over time and over paradigms. Are individuals who are good at learning to perceive novel speech sounds consistently good at it, or does the success of learning fluctuate over time, or with the use of different paradigms? First, we investigate the stability of individual differences over time by assessing performance during five (pre- and post-training) test moments on three separate days with one-week intervals. Second, we investigate the stability over paradigms by comparing the two most commonly used tests of speech sound perception, namely discrimination and identification. 70 native speakers of Dutch participated in a series of training and test sessions, during which they were trained to perceive the Korean three-way lenis-fortis-aspirated contrasts /p-p*-ph/, /t-t*-th/, and /k-k*-kh/, which are difficult for them to distinguish. Results showed, first, that individual differences were very stable over time. Second, the correlation between individuals' discrimination and identification scores was only moderate. Thus, individual differences in learning to perceive novel phonetic contrasts seems to be a stable individual trait over time, but not over paradigms.

11:30

**3aSC10. The effect of sleep on learned sensitivity to a non-native phonetic contrast.** Sayako Earle and Emily Myers (Univ. of Connecticut, 123 Davis Rd., Storrs, CT 06268, frances.earle@uconn.edu)

Consolidation during sleep is thought to play a role in integrating newly learned words into the preexisting lexicon (e.g., Dumay and Gaskell, 2007), while the effect is in stabilizing degraded information against decay when learning to map synthesized speech onto native phonology (Fenn, Nusbaum, Margoliash, 2003). In the current study, we investigated the effects of overnight consolidation on discrimination between new (nonnative) phonetic categories. Fifty-four monolingual English speakers were trained to categorize tokens from a non-native dental-retroflex contrast. Half of the participants were trained in the evening, and the other half were trained in the morning. Discrimination ability was tested 8-14 h post-training and 22-26 h post-training in order that the effects of intervening sleep (evening group) or daytime activity (morning group) could be assessed. Discrimination in the trained vowel context improved after the overnight between-session interval in the night group, but declined slightly in the morning group. Both training groups improved in discrimination ability for an untrained vowel context immediately following the overnight between-session interval, but not before. Results suggest that memory consolidation during sleep, proactive interference during native language exposure, or both play a role in nonnative phonetic learning.



11:45–12:00 General Discussion

# Session 3aSP

## Signal Processing in Acoustics: Acoustical Communications and Sound Source Localization

James C. Preisig, Chair
*WHOI, MS #11, Woods Hole, MA 02540*

### *Contributed Papers*

**8:30**

**3aSP1. Energy efficient transmission policies in non-stationary underwater acoustic channels.** Beatrice Tomasi and James C. Preisig (AOP&E, Woods Hole Oceanogr. Inst., 266 Woods Hole Rd, Woods Hole, MA 02543, btomasi@whoi.edu)

This work focuses on making underwater acoustic communications energy-efficient, by reducing the amount of unsuccessful transmissions. The approach is enabled by *a priori* information regarding the second order statistics of the channel quality. However, the non-stationarity of the physical processes that primarily influence the acoustic propagation makes both channel representation and identification challenging. Therefore, we first evaluate the different types of second order statistics of the underwater acoustic channel measured during two experiments, SPACE08 and KAM11, during which the same source and receiver hardware was employed in different environmental conditions. Then, we classify the different observed second order statistics estimated over a few minutes time intervals and propose suitably trained Markov models to represent the evolution of these different second order statistics. This channel representation is used to derive an optimal transmission scheduling that minimizes the number of transmissions required to deliver a given amount of information B by a given deadline T. In particular, we provide insights on how the structure of the optimal policy changes with the different observed behaviors of the second order statistics of the channel.

**8:45**

**3aSP2. Adaptive orthogonal frequency division multiplexing underwater acoustic communications with limited feedback.** Xiaopeng Huang (Stevens Inst. of Technol., Castle Point on Hudson, Hoboken, NJ 07030, xhuang3@stevens.edu), Aijun Song (Univ. of Delaware, Newark, DE), Walid Ahmed (Stevens Inst. of Technol., Hoboken, NJ), Moshen Badiey (Univ. of Delaware, Newark, DE), and Victor Lawrence (Stevens Inst. of Technol., Hoboken, NJ)

In orthogonal frequency division multiplexing (OFDM) underwater acoustic (UWA) communications, some subcarriers may be subject to deep fading. If the channel state information (CSI) is available at the transmitter side, adaptive transmission techniques (e.g., power allocation) can be applied to mitigate the selective fading effect and increase the overall performance. Therefore, it is more valuable to analyze the UWA channel with limited CSI feedback. In this paper, we adopt two time-varying shallow water acoustic channels (slow-varying environment and fast-varying environment) as examples. Lloyd algorithm is employed to quantize the CSI at the receiver and construct the codebook, which is also known to the transmitter. Simulation results compare the performance between two different channels, and the performance between a few bits of feedback, perfect feedback, and non-feedback, respectively.

**9:00**

**3aSP3. Smart sonic detection and ranging for blind sources localization and separation.** Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, sean_wu@wayne.edu) and Na Zhu (Dept. of Eng. Technol., Austin Peay State Univ., Clarksville, TN)

A new methodology for blind sources localization and separation in arbitrary three-dimensional space is presented. The underlying principle this methodology is the discrete short-time sonic detection and ranging (SODAR) [J. Acoust. Soc. Am. **133**(6), 4054—4064 (2013)] that searches sound sources over discrete frequency-time regions in a systematic and automatic manner. For simplicity yet without loss of generality, only the dominant sound source within each discrete frequency-time region is considered. Once the search is completed, the frequency contents and time instances that correspond to the sound sources identified at the same coordinates are strung together and played back one by one. In this way, one can use N number of sensors to locate and separate S sources, where S can be much larger than N. Numerical simulations and experimental validations of using this methodology for locating and separating arbitrarily time-dependent sound sources in arbitrary three-dimensional space are demonstrated. Advantages and limitations of this methodology are discussed as well.

**9:15**

**3aSP4. Information-theoretic quantification of underwater acoustic source localization performance.** Thomas J. Hayward (Naval Res. Lab., 4555 Overlook Ave SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Metrics that historically have been applied to quantify the performance of signal processing for source localization are algorithm-dependent. For example, performance of conventional beamforming or matched-field processing is usually quantified by main-peak width and secondary-peak levels of the beam response or spatial ambiguity function, while performance of Bayesian localization may be quantified by measures of the statistical dispersion of the *a posteriori* pdf of source location. While algorithm-dependent performance metrics permit comparisons within a given class of signal processing algorithms, they do not provide comparability across algorithm classes. The present work identifies fundamental information-theoretic quantities that can be used as metrics to quantify the source localization performance of diverse signal processing algorithms and thus provide for performance comparisons across signal-processor classes. These quantities include conditional entropy of source location given processor output, mutual information of source location and processor output, and cross-entropy of actual and posterior source-location probability distributions. Applications of these information-theoretic metrics are illustrated in examples of Bayesian localization, conventional beamforming, and matched-field processing of a time-harmonic source in a range-independent shallow-water acoustic waveguide. The results are interpreted in the light of the data processing inequality of information theory. [Work supported by ONR.]

9:30

**3aSP5. Application of information-theoretic performance measures to optimization of array spatial configurations for underwater acoustic source localization.** Thomas J. Hayward (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Information-theoretic measures of acoustic source localization performance provide for performance comparisons that are valid across classes of signal processing algorithms and can be used as performance criteria for the optimization of receiver-array spatial configurations for source localization. This work investigates the use of fundamental information-theoretic quantities, including mutual information of source location and processor output and conditional entropy of source location given processor output, as performance criteria for the optimization of array configurations. Applications of these criteria to the optimization of horizontal and vertical arrays are illustrated in examples of Bayesian localization, conventional beamforming, and matched-field processing of the acoustic field of a time-harmonic source in a range-independent shallow-water waveguide. The optimized array spatial configurations are compared with results obtained using traditional (energy-based) performance measures. [Work supported by ONR.]

9:45

**3aSP6. A relative-entropy approach to distributed passive detection.** Peter C. Mignerey (Acoust. Div., Naval Res. Lab., Peter Mignerey Code 7160, Washington, DC 20375-5350, peter.mignerey@nrl.navy.mil)

There is currently much interest within the ocean acoustics community on using distributed sensor networks to monitor ocean properties. One such task is the application of distributed sensors to passive detection of weak acoustic sources. The joint likelihood detection ratio for a set of distributed sensors leads naturally to the comparison of relative entropy with a detection threshold. As a nondimensional additive measure of information, relative entropy enables data fusion among disparate kinds of sensors, e.g., acoustic and electromagnetic, and mitigates calibration issues. Furthermore, because relative entropy is an integral over probability densities of sensor outputs, it is insensitive to false alarms from transients. In this talk, the theory of relative-entropy detection will be presented, and the method illustrated using acoustic intensity data from hydrophones deployed by the Transverse Acoustic Variability Experiment (TAVEX). For the method to work, accurate estimates of the probability densities for noise and signal intensities must be obtained. For the TAVEX data, superior receiver operating characteristic curves are obtained when the noise and signal distributions are represented by log-normal distributions in comparison with gamma and nonparametric distributions. [Work supported by the Office of Naval Research.]

10:00

**3aSP7. Position estimation of rotating sound source using Kalman filtering based on time difference of arrival measurements.** Jaehyung Lee, Young-Ju Go, and Jong-Soo Choi (Aerosp. Eng., Chungnam National Univ., Yusunggu Gungdong 220, Daejeon 305-764, South Korea, aerojhl@cnu.ac.kr)

In this work, we are interested in tracking a rotating sound source using a Kalman filtering technique based on a set of non-linear time difference of arrival (TDOA) measurements. Array of microphones measure acoustic signal emitted from a rotating source and the TDOA estimates are calculated followed by a solution for hyperbolic position fix. The position estimation of sound source based on TDOA is a popular technique in source localization. The method involves calculation of a set of nonlinear equations and poor accuracy of TDOA estimates often results in inaccuracy in location. In this work, the range difference is expressed by a model movement on which a recursive extended Kalman Filter has been developed. The TDOA measurements optimize the estimated values which are reduced as observation in extended Kalman filtering algorithm. Location estimation is updated from TDOA measurements along with the time history data. The Cramer-Rao Lower Bound (CRLB) is derived and simulations are compared. [Work supported by National Research Foundation of Korea (NRF) grant funded by the Korea government (MEST) (No. 2010-0014978).]



WEDNESDAY MORNING, 4 DECEMBER 2013 PLAZA A, 8:30 A.M. TO 12:00 NOON

## Session 3aUW

## Underwater Acoustics: Contributed Papers in Underwater Acoustics (Poster Session)

Megan S. Ballard, Chair
*Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758*

### *Contributed Papers*

All posters will be on display from 8:30 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:30 a.m. to 10:15 a.m. and contributors of even-numbered papers will be at their posters from 10:15 a.m. to 12:00 noon.

**3aUW1. Acoustic radiation force. Part I: Finite element modeling for elastic objects.** Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com) and Ivars Kirsteins (NUWC, Newport, Newport, RI)

The study of the acoustic radiation force produced by acoustic beams has been the topic of active research in the last few years mainly due to its ability for particle trapping and non-contact manipulations. However, efforts in modeling the radiation force have mainly focused on analytic solutions and thus have been limited to spherical objects, where the acoustic radiation force has been calculated for a plane wave as well as various types of beams. But up until recently, even those efforts were limited to using on-axis beams, where the incident beam is along an axis that goes through the center of the sphere and thus reduces the problem to an axially symmetric one. In this work we use the finite element technique to compute the acoustic radiation force for an arbitrary elastic object and for an arbitrary incident beam. One of the main objectives of this work is to understand and interpret the data we collected at a recent experiment we conducted at the Washington State University test tank facility, where a 2.5-in. PMMA sphere was ensonified by an intense amplitude-modulated ultrasound beam focused at its surface.

**3aUW2. Acoustic radiation force. Part II: Eigenmode excitation of a scaled target.** Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com) and Ivars Kirsteins (NUWC, Newport, Newport, RI)

We describe an experiment that we recently performed at the Washington State University test tank where dynamic acoustic radiation forces were used to excite an eigenmode of a 2.5 in. PMMA sphere in water. The dynamic acoustic radiation force was generated by an intense, amplitude modulated ultrasound signal focused on the surface of the sphere whose amplitude modulation frequency was set to generate acoustic radiation forces modulated at a rate matched to an eigenmode of the sphere. A conventional hydrophone was used to listen to the sphere's acoustic emissions from that particular eigenmode. Unlike conventional acoustic insonification, the dynamic acoustic pressure force is generated by momentum transfer, which creates a mechanical excitation on the object's surface at a rate equal to twice the modulation frequency, analogous to a hammer striking it. In the experiment, the modulated ultrasound beam was scanned horizontally across the sphere using a computer-controlled actuator with the sphere's acoustic emissions measured at each position. To confirm and understand the experimental results, the measured sphere's acoustic emissions were compared to finite element model predictions as a function of horizontal position.

**3aUW3. Test of a towed line array suitable for geoacoustic inversion.** Joel Abdullah, Neil Woodson, Jason D. Sagers, David P. Knobles, Steven A. Stotts, and Thomas Muir (Appl. Res. Labs., Univ. of Texas, ARL:UT PO BOX 8029, Austin, TX 78713-8029, joela@arlut.utexas.edu)

Small line arrays towed by ships may give a unique, cost effective approach for acoustic reconnaissance of seabed characteristics in littoral seas. A 16 element, 1 m spaced, towed line array was designed and developed at ARL:UT for the purpose of ship-towed statistical inference of the seabed. The towed array was tested at the ARL:UT Lake Travis Test Station under a variety of conditions to study array performance. Additional non-acoustic sensors provided information about array shape, stability, and drag as a function of speed, depth, and tail termination. Acoustic data were collected during the test and processed with an adaptive beamformer yielding high SNR, well-localized signals from a fixed 500 Hz source as well as the received signals from small boats of opportunity. The acoustic data are analyzed with a statistical inference approach to estimate the geoacoustic properties of the lake bottom. [Work supported by the ARL:UT internal research and development program.]

**3aUW4. Three-dimensional propagation: Comparison of finite element and coupled-mode solutions.** Megan S. Ballard, Benjamin M. Goldsberry, and Marcia J. Isakson (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

Three-dimensional propagation over an infinitely long cosine-shaped hill is studied using finite element and coupled-mode models. The finite element approach is based on a longitudinally invariant solution technique. The solution is formulated in a Cartesian coordinate system and a cosine transform is applied to eliminate the range-independent dimension. The resulting equation is two dimensional and the solution is calculated for a sufficient range of values of the transform variable. Then the spatial solution is obtained using an inverse cosine transform. The coupled-mode model is formulated in a cylindrical coordinate system, and the solution is obtained using a separation of variables. Modal amplitudes are calculated from the horizontally separated part of the Helmholtz equation using a hybrid technique such that a parabolic solution provides the description of horizontal refraction in the azimuthal direction and a stepwise coupled-mode technique accounts for mode-coupling in the radial direction. The finite element model provides a highly accurate result, limited only by the discretization of the environment and sampling of the cosine transform integral. The coupled-mode solution is approximate, but an examination of the model amplitudes is used to gain insight into the effects of environmental inhomogeneities on the acoustic field. [Work supported by ONR.]

**3aUW5. Ultrasonic measurements of suspended sediment concentrations at Harris Bayou, Mississippi.** Wayne O. Carpenter (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., University, MS 38677, wocarpen@olemiss.edu), Thomas A. Kajdan (Civil Eng., Univ. of MS, Oxford, MS), Bradley T. Goodwiller (Mech. Eng., Univ. of MS, University, MS), Cristiane Q. Surbeck (Civil Eng., Univ. of MS, University, MS), James P. Chambers (Mech. Eng., Univ. of MS, University, MS), Daniel G. Wren, and Roger A. Kuhnle (National Sedimentation Lab., USDA-Agricultural Res. Service, Oxford, MS)

The use of ultrasonic acoustic technology to measure the concentration of fine suspended sediments has the potential to greatly increase the temporal and spatial resolution of sediment measurements while reducing the need for personnel to be present at gauging stations during storm events. In collaboration with the USGS, a customized field deployable system was installed to monitor fine sediment particles, less than 100 micron in diameter, in suspension at Harris Bayou near Alligator, MS. Calibration measurements show good agreement between laboratory grade equipment and the new prototype system. The field unit consists of two immersion ultrasonic transducers measuring attenuation of 20 MHz acoustic signals propagated through suspended particles. The results of field prototype will be presented here.

**3aUW6. Numerical study of the spatial and temporal variability of ambient noise in the coastal region east of Taiwan Strait.** Andrea Y. Chang, Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, yychang@ntu.edu.tw), Sheng-Fong Lin (Green Energy and Environment Res. Lab., Industrial Technol. Res. Inst., HsinChu, Taiwan), and Ruey-Chang Wei (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan)

Noise soundscape represents the characteristics and spatial distributions of the ambient noise level under various noise source mechanisms. This is a significant index while describing an underwater acoustic environment, especially in the subjects related to marine mammal protection. Under the effects of topography, sediment, and oceanographic features, underwater soundscape varies with time and space. This paper focuses on estimating mean soundscape of wind driven noise and shipping noise and their spatial and temporal variability in the coastal region east of Taiwan Strait, which is the main habitat of the Sousa Chinensis. The ambient noise is studied numerically and local wind field and shipping density observed by Automatic identification Systems (AIS) are applied to generate noise source field. As for the ocean environment, the time varying/spatial dependent temperature profiles generated by the Taiwan Coastal Ocean Nowcast/Forecast System (TCONFS), which formulated on the basis of the Princeton Ocean Model, is used for water column variability, and both topography database and geo-acoustic database are used to describe the bottom. The modeling results demonstrate the temporal/spatial variability induced by ocean environment, manifested by measured data. [This work was supported by National Science Council of Taiwan and Bureau of Energy (Grant No.102-D0105).]

**3aUW7. Prototype development of underwater noise impact alert region prediction system.** Yu-Chen Cheng, Andrea Yuan-Ying Chang, Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., No.1, Sec 4, Roosevelt Rd., Taipei 10617, Taiwan, r01525053@ntu.edu.tw), and Sheng-Fong Lin (Inst. of Green Energy and Environment Technologies, Industrial Technol. Res. Inst., Hsinchu, Taiwan)

Offshore wind farms are the main project in western Taiwan. Since the underwater noise generated by piling poses a threat to the marine mammals, the issue of detrimental impact of noise on *Sousa Chinensis* is proposed. To avoid causing behavioral disturbance and injury from pile driving noise, Underwater Noise Impact Alert Region Prediction System (UNIARPS) was established to estimate the acoustic field at any depth and distance from piling source. The system can be illustrated as four components, environment databases, acoustic propagation model, source modeling, and noise level prediction. The ocean numerical model (TCONFS) generates time spatial dependent temperature profiles for water column variability, and geo-acoustic and bathymetry databases are imported as environmental inputs.

Adiabatic mode theory is used to simulate the piling noise propagating in shallow water and the impulsive noise emanating from source is evaluated via finite element method. While the auditory threshold of cetacean set as criterion level, the system can demonstrate the modeling outputs and predict noise impact region, and these results are useful to prior planning on how to station the guarding boats in preventing dolphins entering the noise impact region. [The financial support provided by Bureau of Energy (Grant No.102-D0105) is gratefully acknowledged.]

**3aUW8. Acoustic mode coupling due to subaqueous sand dunes in the South China Sea: Extension of the adiabatic criterion to waveguides with bedforms.** Linus Chiu (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-sen Univ., No. 70, Lienhai Rd., Kaohsiung 80424, Taiwan, linus@mail.nsysu.edu.tw) and Davis B. Reeder (Dept. of Oceanogr., Naval Postgrad. School, Monterey, CA)

The large subaqueous sand dunes on the upper continental slope of the South China Sea (SCS) create a range-dependent ocean acoustic waveguide within which acoustic energy is expected to couple between propagating normal modes. Here, the criterion of adiabatic invariance is extended to the case of a waveguide possessing bedforms. The morphological features of the bedforms modeled in this theoretical and numerical investigation are based on echosounder observations of the SCS sand dune field during a research cruise in the spring of 2012 on the Taiwanese R/V Ocean Researcher 2 (OR2). Using the extended criterion for adiabatic invariance to examine mode propagation over these bedforms, results demonstrate that bedforms increase mode coupling strength such that the criterion for adiabatic propagation is exceeded for waveguides with small bedform amplitude to water depth ratios; increasing bedform amplitude enhances mode coupling. Physically, initially bottom-trapped mode 1 energy abruptly couples to higher adjacent modes, with most of the energy preferentially settling into a few select modes. The scattered energy fills the water column to near-surface depths downrange of the bedforms. Numerical simulations confirm the extended criterion parameterization. [This work was supported by National Science Council of Taiwan.]

**3aUW9. Properties of the Umov vector in shallow water and its dependence on sea surface conditions.** David R. Dall'Osto and Peter H. Dahl (Mech. Eng., UW-Seattle, 914 N 38th St., Seattle, WA 98103, dallosto@u.washington.edu)

In this work, the effects of a rough sea surface on shallow water acoustic propagation are examined using experimental data collected during the ONR sponsored Target and Reverberation Experiment (TREX) off of the coast of Panama City, Florida, in May 2013. The acoustic data were collected from a bottom deployed recording tower that coherently recorded data on a horizontal line array (HLA), a vertical line array (VLA), and on an accelerometer-based vector sensor which was combined with a co-located hydrophone to formulate the Umov vector, or instantaneous intensity vector. The source was lowered from the stern of a research vessel to a depth one-third and two-thirds of the water depth (18 m), and transmitted a multi-frequency pulse from 1 to 4 kHz. These measurements were repeated at positions approximately 10, 20, and 40 water depths away from the tower, along a bearing perpendicular and parallel to the surface wave-crests. During the experiment, the sea surface directional-wave spectrum was measured by a Datawell Waverider buoy moored at the experimental site. Properties of the Umov vector are shown to relate to roughness and directional characteristics of the sea surface. The Umov vector is also studied in relation to the HLA and VLA measurements.

**3aUW10. Depth-tracking of a near-surface target from the deep ocean.** Sheida Danesh and Henrik Schmidt (Massachusetts Inst. of Technol., 143 Albany St., Cambridge, MA 02139, sdanesh@mit.edu)

Determining the depth of an acoustic source in a deep ocean environment can be approached using a variety of existing methods, each with inherent limitations. A method for determining the depth of a moving near-surface acoustic target from a fixed vertical array below the deep ocean critical depth is presented using the characteristics of the Lloyd mirror pattern of a near-surface acoustic signal and a library of calculated patterns. Depth estimates of the moving target are made in real-time and incorporated into a confidence metric for tracking the target motion. Results indicate that this method is robust, performing well in conditions involving environmental mismatch and a moderate amount of surface noise.

**3aUW11. Evaluation of complex broadband biomimetic waveforms for active sonar.** Peter Dobbins (Future Systems, Ultra Electronics Sonar Systems, Leanne House, Avon Close, Weymouth, Dorset DT4 9UX, United Kingdom, peter.dobbins@ultra-sonar.com)

There is an expanding requirement to reduce the impact of man-made sound, including active sonar transmissions, on marine mammals in the defence, offshore, and other sectors. One way this might be achieved in sonar applications is to use signals derived from natural sounds such as the vocalizations of the animals themselves. It might be expected that such sounds would appear more familiar, thus reducing possible abnormal behavioral impacts. This paper reviews the use of such waveforms and presents the results from a trial designed to compare the detection capabilities of a variety of broadband signals, both conventional and 'novel', with a medium frequency active sonar. Two biomimetic signals were tested, one based on sperm whale echolocation clicks and the other on pilot whale whistles. Preliminary analysis suggests the detection performance of these signals using conventional matched filters is comparable with linear FM chirps with a similar bandwidth, but may be improved with detection techniques commonly used for marine mammal vocalizations, such as spectrogram correlation. The paper will conclude with an assessment of the potential impacts of such signals on marine life.

**3aUW12. Underwater navigation using an acoustic spiral wave front beacon.** Benjamin Dzikowicz (Code 7130, Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, ben.dzikowicz@nrl.navy.mil) and Brian T. Hefner (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A spiral wave front beacon consists of an array of transducers which produce a signal whose phase depends on the azimuthal angle at which it is received and a reference signal with constant phase [J. Acoust. Soc. Am. **131**, 3748 (2012)]. A vehicle can determine aspect to the beacon by comparing the phase of the two signals. Progress in the development of this navigation technique will be discussed including results from experiments at Dodge Pond in Connecticut where an unmanned surface vehicle determined its aspect to within 10° by receiving signals from the beacon. Also, tests of a new spiral beacon design [J. Acoust. Soc. Am. **130**, 2506 (2011)] in laboratory and underwater environments will be presented. Underwater experiments are performed at the Navy's Seneca Lake facility in upstate New York using an UUV to record signals. Overall, these results demonstrate that intrinsic phase shifts in the beacon can be handled by signal processing at the receiving vehicle and that the spiral navigation technique is robust in reverberant environments. [Work supported by the Office of Naval Research.]

**3aUW13. Information capacity of an acoustic field in a Pekeris waveguide with an absorbing bottom and spatially correlated surface noise field.** Steven I. Finette and Earl Williams (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

It is well known that the Shannon theory of information sets an asymptotic bound on the maximal rate of transmission of information through a channel with negligible probability of error. This rate, known as the information capacity, can be computed analytically or numerically and numerous results for the capacity in terrestrial communications have been derived. Most analytic results, however, involve free space propagation while in underwater acoustics, the propagation region is spatially bounded. In this presentation, we consider the information capacity in an ocean waveguide comprised of a uniform sound speed in the water column, a penetrable, absorbing bottom and a correlated surface noise field described by the Kuperman-Ingenito source sheet model. An expression for the information capacity of the acoustic field in the water column exterior to a distributed source region is determined by first applying singular value decomposition to the Green's function matrix to obtain independent communication

channels and then using Lagrange multipliers to solve a multiply constrained optimization problem involving the mutual information between source and receive regions. [Work supported by the Office of Naval Research.]

**3aUW14. Supervised machine learning for estimation of rough bottom anisotropy direction using bistatic acoustic scattered fields.** Erin M. Fischell (Mech. Eng., MIT, 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu) and Henrik Schmidt (Mech. Eng., MIT, Cambridge, MA)

An issue when trying to use scattered acoustic fields to classify underwater targets is the strong directional scattering due to anisotropic rough bottom structure that occurs in the 1-5 kHz frequency range. Autonomous Underwater Vehicles (AUVs) are uniquely suited to exploit this three dimensional field with the goal of estimating the anisotropy direction of the bottom roughness. Estimation of the angle of bottom roughness ridges relative to an acoustic source is carried out using a combination of supervised machine learning and AUV behaviors. Anisotropic Goff-Jordan power spectrum bottom scattered fields in 15 degree angle increments are generated using SCATT and OASES acoustic packages. The amplitudes of these fields are sampled into sets of 5-20 AUV waypoints. Support Vector Machine (SVM) regression is used to train a model, and an independent test data set is used to evaluate the validity of the model. A confidence model is constructed and critical waypoints identified using the test set results. The confidence and SVM models can then be used to determine bottom roughness angle in real time, as demonstrated using the LAMSS MOOS-IvP simulation environment.

**3aUW15. Generation and propagation of oceanic T-waves using elastic parabolic equation solutions.** Scott D. Frank (Mathematics, Marist College, 3399 North Ave.,Poughkeepsie, NY 12601, scott.frank@marist.edu), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), and Robert I. Odom (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Oceanic T-waves are essential for location and identification of seismic sources since they travel long distances in the ocean and are typically the largest signals received at hydrophone arrays or coastal monitoring stations. T-waves either link directly into the SOFAR channel by conversion of elastic wave energy at a downward sloping interface between the elastic and fluid media, or are generated when elastic energy couples into low order acoustic modes due to bathymetric inhomogeneities. Elastic parabolic equation solutions will demonstrate generation and long range propagation of oceanic T-waves in the water column when the source is located in an elastic ocean bottom. Elastic parabolic equation solutions will be used to describe effects of ocean bottom parameters on transmission characteristics of a sloping boundary. The impact of small-scale bathymetric changes, for example due to a rough ocean bottom, will be characterized by averaging acoustic wavenumber spectra resulting from multiple bottom realizations. Favorable characteristics for T-wave generation will be determined. The impact of large scale bathymetry changes, such as a seamount or underwater ridge, will also be discussed. [Work supported by ONR.]

**3aUW16. Information transfer of broadband sonar echoes.** Charles F. Gaumond (Acoust. Div., US Naval Res. Lab., CODE 7162, 4555 Overlook Ave. SW, Washington, DC 20375, charles.gaumond@nrl.navy.mil)

The ability to compute information transfer of a broadband signal corrupted with additive noise is straightforward, but the case of signal corrupted by convolution with a stochastic, propagation impulse-response is more complicated. The known, integral solution is presented along with numerical methods for estimating differential entropy and mutual information using kernel density estimators. Numerical examples using simulations of echo and propagation responses are shown. Results are shown using echo data from Clutter09, an ocean experiment performed with an echo repeater. These echoes were generated with a known, modeled target response. The mutual information between the modeled target response and the echoes, which propagated through ocean environments, is also shown. These results are discussed with respect to developing a method of obtaining an information sonar equation that could be used to estimate the sonar parameters required to perform an information based task, such as signal classification. [Research funded by the Office of Naval Research.]

**3aUW17. Clutter statistics of long-range wideband echoes from fish aggregations off the Oregon coast.** Roger C. Gauss, Joseph M. Fialkowski (Acoust. Div., Naval Res. Lab., Code 7164, 4555 Overlook Ave., S.W., Washington, DC 20375-5350, roger.gauss@nrl.navy.mil), and Richard H. Love (BayouAcoust., Abita Springs, LA)

Echoes from fish can be the dominant source of reverberation over a range of important sonar frequencies and grazing angles. Moreover, fish echoes from broadband signals often retain coherent structure (generate clutter) after undergoing normalized match-filter processing. Coupled with their inherent spatiotemporal variability, fish can thus be a significant clutter problem for active sonars. Using a towed source and horizontal line-array receiver, measurements of mid-frequency (1.5-11 kHz) backscattering from aggregations of fish were made from the R/V New Horizon in five shallow-water and shelf-break areas off the coast of Oregon (Astoria Canyon to Heceta Bank) during July and August 2012. The experiment and the frequency-dependent echo statistics in relation to the observed distribution and behavior of the two primary resident fish species (Pacific hake and Pacific sardines) are discussed. For example, the short-time echo variability and spatial patchiness were characteristic of the mid-water (hake) and near-surface (sardine) fish observed concurrently on echosounder displays. Furthermore, the clutter's probability density functions were found to be non-Rayleigh but well modeled by NRL's Poisson-Rayleigh clutter model that provides a physical context for relating data distributions to scatterer attributes. [Work supported by the Office of Naval Research.]

**3aUW18. Interpulse noise assessment during shallow water seismic survey using airgun excitations.** Shane Guan (National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Joseph F. Vignola, and Teresa J. Ryan (Dept. of Mech. Eng., Catholic Univ. of America, Washington, DC)

Offshore energy exploration and geophysical research activities using seismic airgun arrays are known to generate intense underwater sound. Such seismic imaging has the potential to impact marine mammals through hearing impairment and behavioral modification. Few studies have investigated multipath propagation and reverberation from these airgun impulses. These phenomena could increase the duty cycle within the sound field sufficiently to impact long distance communication or result in detrimental acoustic masking for marine mammals. We report initial findings on the elevation of the sound level between airgun impulses during a shallow, open-water seismic survey. This work uses continuous recordings collected from three bottom-mounted hydrophones deployed in the Beaufort Sea in summer 2012. A quantitative method is used to examine the root-mean-squared noise levels between seismic impulses and the noise level dependence on source range. Preliminary results show that ambient noise increases above non-impulsive harassment levels defined by the National Marine Fisheries Service (120 dB re: 1 microPa) for a portion of the time between seismic impulses at intermediate ranges from the source. In addition, the duration of reverberation is related to the source range, with significantly longer decay times measured on hydrophones at greater distances from the source.

**3aUW19. Dual-channel orthogonal modulation differential pattern time delay shift coding underwater acoustic communication method.** Xiao Han, Jingwei Yin, Xiao Zhang, and Chi Wang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Bldg., Nantong St., Nangang District, Harbin, Harbin 150001, China, hanxiao1322@hrbeu.edu.cn)

Information is carried by the time delay between adjacent code elements in differential pattern time delay shift coding system. It has an ability of anti inter-symbol interference and the Doppler effects. In order to obtain higher communication rate and reduce the interference between channels, this paper proposes a dual-channel orthogonal modulation differential pattern time delay shift coding underwater acoustic communication method and selects the balance Gold sequence as Patterns. At the transmitter end, divide input bits into two channels to differential pattern time delay shift encode and modulate

orthogonally. Then add the encoded signal of two channels together to transmit. At the receiver end, demodulate the received signal orthogonally, search the correlation peak position of patterns. And estimate time delay to restore the original input information. Dual parallel channel transmission mode effectively improves communication rate of differential Pattern time delay shift coding and the orthogonal modulation method greatly reduces the interference between channels. Simulation research is carried on for the method, at transmission data $2x10^{4bit}$ and SNR -5dB, information is recovered at a communication rate 205 bit/s with very low bit error rate.

**3aUW20. Propagation effects of surface waves in two unperturbed mode models.** Frank S. Henyey and Eric I. Thorsos (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, frank@apl.uw.edu)

Two unperturbed mode simulation models for acoustic propagation with a wavy surface have been developed. The first, LIN, is the first order in the surface elevation for a fixed number of modes. It is the model that provides the starting point for deriving transport equations, as the mode coupling spectra needed for transport theory are practical to evaluate. The second model, DAE, is more accurate for a larger number of modes, but the mode coupling spectra are very difficult to evaluate. The results are compared for an example propagation environment, and consequences to the accuracy of the transport theory are discussed.

**3aUW21. Three-dimensional primary acoustic field characterization for a seismic airgun array.** Arslan M. Tashmukhambetov, George E. Ioup, Juliette W. Ioup (Phys., Univ. of New Orleans, New Orleans, LA), Natalia A. Sidorovskaia (Dept. of Phys., Univ of Louisiana at Lafayette, Lafayette, LA), Joal J. Newcomb (Naval Oceanogr. Office, Stennis Space Ctr., MS), James M. Stephens, Grayson H. Rayborn (Phys. and Astronomy, Univ. of Southern Mississippi, Hattiesburg, MS), and Phil Summerfield (Geodetics & Cartography, ExxonMobil Corp., UIT, Houston, TX)

The Littoral Acoustic Demonstration Center conducted the Source Characterization Study in 2007 (SCS07) to measure the 3-D acoustic field of a seismic airgun array in the Gulf of Mexico. Three moorings with sensitive and desensitized hydrophones at different depths were deployed as well as hydrophones suspended from a ship, while a seismic source vessel shot specified lines. Hydrophone positions were measured. Peak pressures, RMS sound pressure levels (SPL), sound exposure levels, total shot energy spectra, one-third octave band analyses, and source directivity studies are used to characterize the field. Summary results are first calculated for each hydrophone. These are then combined to give isopleths for azimuthal cuts at 0, 45, 90 degrees, etc., for the spatial domain measures. Plots for each solid angle bin give these and frequency measures analyzed versus range. Zero-to-peak pressures directly under the array go from 210 dB for depths less than 200 m down to 195 dB at 1200 m. At 2000 m horizontal range the pressures go from 160 dB near the surface to 175 to 180 dB at 1200 m. RMS SPL is about 5 dB smaller. [Research supported by the Joint Industry Programme through the International Association of Oil and Gas Producers.]

**3aUW22. Multi-static scattering characteristics of submerged objects with experimental investigation.** Yoon Hee Ji, Gi Hoon Byun, Jea Soo Kim (Ocean Eng., Korea Maritime Univ., Dong-Sam dong, YoungDo Gu, Busan 606-791, South Korea, 1002wine@hanmail.net), Ho Seuk Bae, and Woo Shik Kim (Agency for Defense Development, Changwon, South Korea)

The scattering characteristics of target echoes are essential for detecting and classifying the submerged objects. The target strength, which is widely used in mono-static sonar system, is also important in multi-static sonar system to identify the submerged target. In this presentation, a series of experiments in the acoustic water tank was conducted to measure the target echoes from submerged cylinder-shaped target with multi-static measurement system, which consists of a single transmitter and 16 receivers. The target strengths are presented in 2-dimensional plane as a function of receiver position according to target aspect angle. The numerical simulation results based on Kirchhoff approximation are presented to explain some characteristics of the measured multi-static target echoes. [Work supported by Agency for Defense Development, Republic of Korea.]

**3aUW23. Fluctuations of arriving narrowband signal's direction in horizontal plane in shallow water.** Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru) and Valery Grigorev (Phys., Voronezh Univ., Voronezh, Russian Federation)

Directions of amplitude (envelope) and phase fronts in horizontal plane of signals with some width of spectrum coming to receiving array in shallow water are studied. As an example experiment Shallow Water 2006 is analyzed, where LFM signals of the frequency $300\pm30$ Hz were used for acoustical sounding on the distance ~20 km. Visible length of horizontal part of L-shaped array was ~200 m. It was shown that fluctuations of direction of phase and amplitude fronts took place with angle between them about $2.5^\circ\pm1.5^\circ$ were registered. Results are interpreted as manifestation of frequency dependence of horizontal refraction initiating variation of horizontal angle of coming rays from pulse to pulse during time of observation. [Work was supported by RFBR and BSF.]

**3aUW24. Estimation of level-crossing rate of signals reflected by ocean surface based on rough surface scattering theory.** Joonsuk Kim (Dept. of Elec. and Electron. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Rm. 381B Yonsei Eng. Res. Park, Seoul 120-749, South Korea, kimjs1st@yonsei.ac.kr), Il-Suek Koh (Dept. of Electron. Eng., Inha Univ., Incheon, South Korea), and Yongshik Lee (Dept. of Elec. and Electron. Eng., Yonsei Univ., Seoul, South Korea)

A method is proposed for theoretical estimation of the level-crossing rate of the underwater acoustic communication signals that are reflected from the ocean surface. The variation in the reflection coefficient of the moving ocean surface causes the intensities of the received signals to fluctuate. In this work, the reflection coefficient is obtained by modeling the time-varying characteristics of the ocean surface based on rough surface scattering theory. The surface of the ocean is modeled by Gaussian processes that are characterized by ocean spectra such as the Pierson-Moskowitz and Durden-Vesecky spectra. Furthermore, Gaussian random functions with a particular correlation time are incorporated to model the surface that continuously varies over time. Then the standard Periodogram analysis is applied to estimate the autocorrelation function. Finally, the level-crossing rate is calculated with the negative curvature of the autocorrelation function. For verification, comparison of the simulated results with the measured data is provided.

**3aUW25. Effects of wideband source types on accuracy of wideband path-loss prediction based on finite-difference time-domain scheme.** Yongjune Kim (Elec. and Electron. Eng., Yonsei Univ., C133, Eng. Bldg. C, 134, Shinchon-dong, Sudaemoon-ku, Seoul 120-749, South Korea, sum57@yonsei.ac.kr), Il-Suek Koh (Electron. Eng., Inha Univ., Incheon, South Korea), and Yongshik Lee (Elec. and Electron. Eng., Yonsei Univ., Seoul, South Korea)

The finite-difference time-domain (FDTD) scheme is a well-known numerical algorithm that can solve a wideband response of a wave equation in time domain. When the FDTD scheme is applied to the problem of the underwater path-loss, however, the wideband source excitation may generate various problems at low frequency band. For instance, when a source is implemented with a large magnitude near the zero frequency, a spurious response can be generated in the FDTD simulation due to the DC component that cannot propagate. On the other hand, when a source is implemented with very small magnitude at low frequencies, the numerical accuracy cannot be guaranteed. In this work, implementation of a new wideband source for FDTD scheme is proposed that is suitable for response over a wide bandwidth, including very low frequencies. The Tukey window is applied to a wideband Gaussian pulse in the frequency domain, which eliminates the DC component effectively. By utilizing the steep slope of a Tukey window, higher accuracy is achieved in the low frequencies. To verify the proposed wideband source, the path-loss results based on the FDTD scheme are compared with the famous normal mode solution, KRAKEN, as well as the ray solution, BELLHOP.

**3aUW26. A new type flextensional transducer.** Yu Lan, Wei Lu, Yongjie Sang, and Kuan Li (Harbin Eng. Univ., Nantong St. No.145, Hei Longjiang Province, Harbin, Harbin 0086, China, lanyu_2013@126.com)

In a field of an oceanographic survey, low frequency sound wave is often used because of low attenuation and good propagation characteristic in water. It is well known that flextensional transducer is a typical low frequency underwater source for oceanographic research, utilizing flexural vibration to realize low frequency radiation with small size. However the size of traditional flextensional transducer becomes relative large when the frequency decreases to a few hundred Hertz, and it is still a difficult problem to solve. A new type of class IV flextensional transducer was proposed, and original piezoelectric ceramic stacks were replaced by three groups of small class IV flextensional transducers, resulting in frequency decrease and volume displacement expansion. Make use of ANSYS finite element software, the new class IV transducer was modeled and analyzed. The experimental data showed the frequency of this new transducer design obviously decreased compared with traditional IV flextensional transducer at the same size. Key words: Flextensional transducer; low frequency; small size; Finite Element Method

**3aUW27. Sensitivity of the underwater sound field in submarine canyons to water column variability.** Ying-Tsong Lin, Weifeng Gordon Zhang, and Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Bigelow 213, M.S.#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

The ocean dynamics in the geologically and morphologically complex submarine canyons can have strong spatial and temporal variability due to the presence of internal tides/waves and upwelling currents. Our fundamental research question is what the acoustic effects of these oceanographic processes are in such environments where the sound propagation is also strongly influenced by the seafloor complexity. A simple example showing the joint effects is the bottom reflection of sound that has complicated patterns depending on the shape of canyon seafloor, seabed properties, and acoustic incident angles. Among these factors, the incident angle is the link connecting the acoustic effects of marine geology and physical oceanography. Specifically, the ocean dynamics changes the water column stratification and thus the incident angle of sound onto the seafloor, which explicitly determines the reflection of sound from the complex canyon seafloor. More involved examples using integrated regional ocean and full-field sound propagation models will be shown in the talk, and sensitivity analysis of underwater sound propagation along and across canyons will be performed. [Work supported by the Office of Naval Research.]

**3aUW28. Computational modeling of acoustic wavefronts propagating in an underwater environment with uncertain parameters.** Sheri Martinelli (202 Rochambeau Ave., 1176 Howell St., Newport, RI 02841, sheri_martinelli@alumni.brown.edu)

High frequency simulation of underwater sound propagation is a vital part of modeling and simulation of acoustic systems for evaluation and performance prediction. Existing simulations use deterministic ray tracing to propagate the acoustic field and simulate uncertainty by varying results according to basic distributions (e.g., adding ”jitter” to ray arrival angle). Rather than incorporate randomness as a form of post-processing, this work seeks to model uncertainty where it exists in the underwater environment where it is easier to specify, and then propagate the relevant random quantities through the system applying stochastic collocation to an existing deterministic model. Further, this work addresses the drawbacks of ray tracing by taking the deterministic method to be a model that computes propagation of entire wavefronts rather than rays, thus maintaining error control over the physical domain. To this end, generalized polynomial chaos expansions are applied to a level-sets based wavefront propagation method to model the effects of uncertain parameters in an underwater environment. This approach allows for not only simple extraction of the process moments, but also yields an expression for the wavefronts in terms of random variables which can readily be simulated. [Work supported by ONR.]

**3aUW29. Modeling the generation and propagation of hydrodynamic hull noise near the ocean surface.** Rob Doyle (Explosion and Fluid Dynam., Martec Ltd., 5189 South St., Apt. 4, Halifax, NS B3J 1A2, Canada, rob.doyle@dal.ca), Mae Seto (Manned and Unmanned Systems for Mine Defence, DRDC Atlantic, Halifax, NS, Canada), and Julio Militzer (Mech. Eng., Dalhousie Univ., Halifax, NS, Canada)

Hydrodynamic hull noise is an important consideration for determining the detection envelope of SONAR domes mounted to surface vessels. In order to model the generation of this noise by a moving ship, a hybrid computational hydro-acoustic modeling methodology has been developed by combining the Lighthill-Curle acoustic analogy with the Numerical Wind Tunnel computational fluid dynamics code. This model has been shown in previous work to significantly over-predict the experimentally observed far field sound of Canadian Forces Auxiliary Vehicle Quest in at-sea acoustic trials. This deficiency is shown to be due in part to neglecting the Lloyd’s Mirror interference effect of the sea surface in the Lighthill-Curle equations. By utilizing a method of images solution to the acoustic analogy to simulate the Lloyd’s Mirror interference, an average sound pressure level improvement of 25 dB was obtained. This solution is compared and contrasted to a model utilizing the Lloyd’s Mirror interference of a simple source, and a normal mode sound propagation model, and is shown to be superior for transmission ranges up to 2 km.

**3aUW30. Near- and far-field simulations of coherent backscattering from scatterers in finite sized aggregations.** Adaleena Mookerjee and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., 2010 Autolab, Ann Arbor, MI 48109, adaleena@umich.edu)

Classification of scatterers is a difficult but important step in active sonar applications. Active sonar signals in an ocean environment are scattered by surface roughness, and volume inhomogeneities in the bottom and water column. Fish schools may be important water-column clutter sources and, under some circumstances, may preferentially backscatter sound because of acoustic coherent backscattering enhancement (CBE). Here, the addition of in-phase scattered waves from propagation path pairs can readily explain a scattered intensity enhancement of a factor of two in the direction opposite to that of the incident wave. This presentation describes CBE simulations for finite sized aggregations of point scatterers using the Foldy (1945) equations that show much larger enhancements are possible in the far-field of the scattering aggregation. The simulations are validated in the near field with the theory from Akkermans *et al*. (1986), and with existing CBE optics and acoustics experiments from Wolf and Maret (1985) and Aubry *et al*. (2007). The dependence of the width of the CBE backscattered peak in the far-field is reported. Extension of these results to sonar pulse scattering from schools of fish is also very briefly discussed. [Work supported by the Office of Naval Research.]

**3aUW31. Modeling of underwater noise from pile driving using coupled finite element and parabolic equation model with improved parabolic equation starting field.** Jungyong Park, Woojae Seong, and Keunhwa Lee (Dept. Ocean Eng., Seoul National Univ., Seoul, South Korea, ioflizard@snu.ac.kr)

An offshore wind farm will be constructed in the Yellow Sea, west of Korean Peninsula, where there are extensive fishing activity and numerous fishery farms. To study the effect of underwater piling noise on fishing and marine lives, we model the pile driving noise propagation using coupled FE and PE model. The near-field noise is computed by FE model, considering detailed specifications of the pile driving system. We apply 2D axis-symmetric geometry and utilize acoustic structure interaction analysis in the frequency domain. The FE results are used to compose the starting field for PE model, where appropriate range selection is an important factor to cover most of the contributing ray paths. Extrapolation technique to compensate the lack of FE data and the numerical filtering method to smooth the FE result are discussed. In the far-field, the noise propagation is modeled by the split step Pade PE algorithm. The improved PE starting field seems to give refined result than previous coupled model.

**3aUW32. Directionality of ambient noise measurements in Barrow Strait of the Canadian Arctic.** Nicos Pelavas, Sean Pecknold, Carmen E. Lucas, and Garry J. Heard (DRDC Atlantic, 9 Grove St., Dartmouth, NS B3A 3C5, Canada, nicos.pelavas@drdc-rddc.gc.ca)

In August 2012, a field trial was carried out in Barrow Strait south of Gascoyne Inlet in the vicinity of 74.630 N 91.340 W. Underwater acoustic data was collected using a JASCO Autonomous Multichannel Acoustic Recorder (AMAR) and in-house designed sensor systems called Starfish Cubes. The Starfish Cubes were deployed twice, at different locations, each for one week duration and at depths of approximately 110 m. The Cubes consist of seven hydrophones with 1 m spacing and geometrically configured as three cross-dipoles with a central hydrophone, and have an operational frequency range of 5—750 Hz. During the trial 400 and 500 Hz tones were transmitted from discrete locations at various ranges. By using a beamforming method the tones were used to determine the orientation of the Starfish Cubes during their data collection periods. This enables investigation of the horizontal and vertical directionality of ambient noise. Unique localized sources contributing to the ambient noise are discussed such as a nearby grounded iceberg and a low frequency wandering tonal.

**3aUW33. Scattering objects imaging using an autonomous underwater vehicle towing a source and an horizontal array.** Samuel Pinson and Charles W. Holland (Penn State Univ., Appl. Sci. Bldg., Rm. 202a, State College, PA 16802, samuelpinson@yahoo.fr)

Recently, seabed sound-speed profile measurement by the image source method had been performed using an Autonomous Underwater Vehicle (AUV) towing a broadband source (frequency band from 1600 to 3500 Hz) and a linear array of hydrophones. This method provides an automatic process by the use of the semblance function. In that communication, the semblance ability to detect coherent reflections is used to image scattering objects such as mud volcanoes by integrating the results of successive measurement along the AUV track. Due to the horizontal array configuration, there is an ambiguity on the scatterers localization that could be solved by the use of hydrophone triplets.

**3aUW34. Backscattering spectrum of a solid cylinder next to a horizontal surface when the cylinder's axis is not horizontal.** Daniel Plotnick, Phillip L. Marston (Phys., Washington State Univ., 1510 NW Turner DR, Apt. 4, Pullman, WA 99163, dsplotnick@gmail.com), Aubrey Espana, and Kevin L. Williams (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

When a solid cylinder lies proud on horizontal sand sediment there has been progress in understanding the backscattering spectrum as a function of grazing angle and the viewing angle relative to the cylinder's axis [Williams *et al*., J. Acoust Soc. Am. **127**, 3356-3371 (2010)]. The resulting evolution of the target strength spectrum is sometimes referred to as the "acoustic color" or the "acoustic template." For cylinders having identical ends and a transducer at a fixed grazing angle relative to the cylinder's center, viewing the cylinder over a 90 degree range is sufficient for characterizing the template. If the cylinder's axis has a vertical tilt such that one end is partially buried in the sand, then the symmetry of the template is altered and a 180 degree range is required. Some of the changes in the template can be approximately modeled using a combination of geometrical and physical acoustics. The resulting analysis gives a simple approximation relating certain changes in the template with the vertical tilt of the cylinder. A similar approximation also applies to a metallic cylinder adjacent to a flat free surface and was confirmed in tank experiments. [Work supported by ONR.]

**3aUW35. Some results from the very shallow water TREX13 reverberation experiments using the Five Octave Research Array triplet module.** John R. Preston (ARL, Pennsylvania State Univ., P. O. Box 30, M.S. 3510,State College, PA 16804, jrp7@arl.psu.edu)

A large experimental effort called TREX13 was conducted in April-May 2013 off Panama City, Florida. As part of this effort, reverberation and clutter measurements were taken in a fixed-fixed configuration in very shallow water (~20 m) over a 22 day period. Results are presented characterizing reverberation, clutter, and noise in the 1800-5000 Hz band. The received data are taken from the triplet sub-aperture of the Five Octave Research Array (FORA). The array was fixed 2 m off the sea floor and data were passed to a nearby moored ship (the R/V Sharp). An ITC 2015 source transducer was fixed 1.1 m off the seafloor nearby. Pulses comprised of gated CWs and LFMs were used in this study. Matched filtered polar plots of the reverberation and clutter are presented using the FORA triplet beamformer. There are clear indications of biologic scattering. Some of the nearby shipwrecks are clearly visible in the clutter, as are reflections from a DRDC air-filled hose. The noise data show a surprising amount of time-dependent anisotropy. Some model-data comparisons are made using the author's normal mode based reverberation model. Help from the Applied Physics Laboratory at the University of Washington was crucial to this effort. [Work supported by ONR code 322OA.]

**3aUW36. Eigenspace dynamics of sample covariance matrices.** Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., A405, Victoria, BC V8P 5C2, Canada, jorgeq@uvic.ca) and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR)

Estimation of the sample covariance matrix is a challenge in array processing, particularly with large-aperture arrays operating in dynamic environments affected by fast maneuvering interferers and background noise. Minimizing the impact of time-dependent variations in the underlying signal statistics requires short observation intervals, thereby reducing the number of snapshots available for covariance estimation. Distinguishing between true variations in the received signal statistics and artifacts introduced by insufficient samples is still an open field of research. Recent developments in random matrix theory (RMT) have provided mathematical foundations to understand the behavior of sample eigenvalues and eigenvectors, and how they deviate from their population counterparts due to the lack of snapshot support. Similarly, expressions have been obtained to describe the "distance" between an initial eigenspace (spanned by p eigenvectors), relative to a subsequent eigenspace (spanned by q eigenvectors). In this paper, simulations corresponding to a horizontal array operating in a realistic environment are used to investigate RMT-based metrics that quantify time-dependent eigenspace stability. This research develops mathematically justifiable methods for proper data segmentation into intervals that exhibit local stationarity, providing data-driven higher bounds for the number of snapshots available for the computation of sample covariance matrices.

**3aUW37. Passive ranging in strongly range-dependent environments: Effects of mode coupling on the waveguide invariant.** Alexander W. Sell (Acoust., Penn State Univ., 830 Cricklewood Dr., Apt. 207, State College, PA 16803, aws164@psu.edu) and R. Lee Culver (Acoust., Penn State Univ., University Park, PA)

Prior work has shown that the value of the shallow water waveguide invariant changes in range-dependent environments due to non-uniform phase and group speed along the propagation path caused by either a range-dependent bathymetry or sound speed profile. Much of the work on these scenarios has dealt with weak range-dependence and the effects of mode coupling were neglected. In certain situations when mode coupling occurs, energy from higher order, surface-reflected-bottom-reflected modes may be lost to lower order, surface-refracted-bottom-reflected modes. These lower order modes, which do not interact with the surface, are associated with waveguide invariant values that differ greatly from the standard shallow water approximation where the waveguide invariant equals one. This talk will examine a case from the 2007 CALOPS experiment where the adiabatic approximation for modal propagation is no longer valid and mode coupling appears to be important. Acoustical data and analysis will be presented to demonstrate the effect that mode coupling has on the waveguide invariant in strongly range-dependent environments, and methods for incorporating coupled modes into waveguide invariant estimates will be discussed. [This research was supported by the Applied Research Laboratory, at the Pennsylvania State University through the Eric Walker Graduate Assistantship Program.]

**3aUW38. Multistatic sound speed profile estimation.** Hisashi Shiba (Radio Application Div., NEC Corp., 1-10, Nisshin-Cho, Fuchu, Tokyo 183-8501, Japan, h-shiba@aj.jp.nec.com)

Sonar is an indispensable component of harbor security systems. Sound propagation is one of the big problems for coverage estimations, since sound speed profiles are complicated under the complex environment like harbors. A new approach for sound speed profile estimation had been proposed using surface scattering by single sonar for frequent measurements which are required in the operation planning phase of high accuracy coverage evaluations. Although the configuration is simple, it consumes much time for higher accuracy estimations, because multiple angle transmissions need multiple waiting time and averaging considering surface fluctuations also need lots of time. This requirement is not favorable for quick sonar operations. One of the ideas reducing estimation time is using multiple sonars under multi-static configurations. This new concept uses multiple sonars, and it is suitable for harbor protections, since all sonars on the bottom do not become obstacles for ship navigations unlike tethered arrays used in acoustical tomography. And it can be said that multiple sonars should be deployed for covering wide and complex harbor areas with networks. Under this situation, a multi-static operation is a natural conclusion for harbor security. This new approach reduces the total estimation time. The rest problem is accelerating computing time of solving nonlinear simultaneous equations. A new acoustical structure model is under evaluation. The trial results are reported, if it is found to work well.

**3aUW39. Measurements of the peak pressure and sound exposure level from underwater explosions.** Alexander G. Soloway (Dept. of Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, soloway@u.washington.edu) and Peter H. Dahl (Dept. of Mech. Eng. and Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

There is an interest by the Navy to determine the sound field produced by underwater explosions to minimize the impact on marine life during training exercises. This work presents measurements of underwater explosions collected 7 km off the coast of Virginia in shallow water (depth 14 m) with sound speed conditions considered approximately iso-speed. Explosive charges with TNT equivalent weight 0.1 to 6.0 kg (W) were deployed at approximately mid-water and bottom depths. Acoustic data were recorded using a 9 element vertical line array at range 430 m and single-element autonomous systems at ranges 170, 430, and 950 m. The peak pressures and sound exposure levels (SEL) are calculated from the data; at 430 m peak pressures as high as 220 dB re 1 $\mu$Pa and SEL as high as 190 dB re 1 $\mu\text{Pa}^2$ s were measured. The peak pressures are compared to semi-empirical equations that are functions of range and W to the one-third power, such as Arons [J. Acoust. Soc. Am. **26**, 343-346 (1954)], and both the peak pressures and SEL are compared to simulations obtained using the parabolic wave equation. [Research supported by Naval Facilities Engineering Command.]

**3aUW40. Surface wave shape inversion from forward scattered ocean acoustic data.** Sean Walstead and Grant Deane (ECE/SIO, UCSD, 9500 Gilman Dr., 0407, La Jolla, CA 92093-0407, swalstead@ucsd.edu)

Prior work has shown that surface wave shape can be determined by analyzing underwater surface reflected acoustic signals in a wave tank. In this talk, forward scattered data from the Surface Processes and Communications Experiment (SPACE08) is analyzed with regard to surface wave shape inversion. Multipath arrivals representing surface, bottom-surface, and surface-bottom paths are distinguishable, implying that knowledge of the surface is known approximately 1/3, 1/2, and 2/3 the distance between source and receiver. Surface scattering losses including out of plane scattering and small scale roughness are numerically simulated and compared to actual ocean data. Methods are proposed for including these loss factors in a forward model of surface scattering that can be correlated with environmental conditions observed at the Martha's Vineyard Coastal Observatory.

**3aUW41. Analysis of ambient noise in the habitat of Indo-Pacific humpback dolphin (*Sousa chinensis*) in the West Coast of Taiwan.** Ruey-Chang Wei, Lian-Han Kuo (Inst. of Appl. Marine Phys. and Undersea Technol., National Sun Yat-sen Univ., Kaohsiung, Taiwan), Jeff Chih-Hao Wu, and Chi-Fang Chen (Dept. of Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei, Taiwan, d98525001@ntu.edu.tw)

The west coast of Taiwan is one of the major habitats of Indo-Pacific humpback dolphin (*Sousa chinensis*). Ambient noise, changes with natural environment and human activities, is possible to affect the behaviors of marine mammals. Thus, it is necessary to conduct a long-term and systematic investigation of ambient noise in this area. This study deployed two underwater acoustic recorders (SM2M) in New Huwei River of the Yun-Lin coastal area (site YL) and Waisanding sandbar (site WS). 68- and 45-day acoustic data were collected in site YL and site WS. Results show that the low-frequency noise in site WS is lower than site YL due to the contributions of shipping or mechanical noises. In site YL, ambient noise of 1 to 2 kHz contains periodic changes because of the behaviors of croakers. Croakers usually appears before midnight in site YL, but no similar phenomenon is found in site WS. The frequency overlap between hearing range of marine mammal and high-level ambient noise is possible to cause the masking effect, even hearing loss. [Sponsored by the Forestry Bureau, Council of Agriculture, Taiwan under project "Population Ecology of Chinese White Dolphins and Ambient Noise Monitoring in its Habitat" No. 101-08-SB-14).]

**3aUW42. The effects of source motion and bottom bathymetry on temporal coherence in shallow water propagation.** Jennifer Wylie and Harry DeFerrari (Appl. Marine Phys., Univ. of Miami, RSMAS, 4600 Rickenbacker Cswy., Ops 11, Key Biscayne, FL 33432, jennie.wylie@gmail.com)

Previous studies on coherence have been focused on the effects of water column fluctuations on temporal coherence with a stationary source/receiver setup. However, with focus being turned to moving source setups there has been documented a significant drop in temporal coherence. With a moving platform, the propagation path will change based on relative source/ receiver position, and hence the bathymetry along the path will vary. Here, we will examine the effects that this bathymetric variation and related ship speed contribute to coherence loss. A range dependent parabolic equation model will be used to predict the temporal coherence for individual mode arrivals. A slowly varying random bottom will be introduced to the model and the coherence calculated for different ship speeds and for both radial and tangential tracks. Results will be compared with stationary source/receiver setups in order to determine at what ship speed/ bottom bathymetry does source motion become the driving factor in loss of coherence versus water column fluctuations from a stationary setup. Preliminary results indicate that at a speed of 2 knots, there is remarked loss of coherence at all modes except the first with even small variations in bottom bathymetry, which is in agreement with experimental results.

**3aUW43. Research on phase generated carrier demodulation algorithm phase drift of fiber-optic hydrophone.** Ge Yu and Jinshan Fu (College of Underwater Acoust. Eng., Harbin Eng. Univ., Bldg. 145, Nantong St., Harbin, Harbin 150001, China, liz.221@163.com)

The paper outlines phase generated carrier (PGC) modulation and demodulation principles of interferometric fiber-optic hydrophone. For demodulation error caused by phase drift, the paper proposes a simple and practical method, using the cross-correlation between the measured signal and the two reference signals to lock the small signal of desired frequency and measure it. Two reference signals with a constant phase difference can directly output the phase of the measured signal. The paper realizes a real-time PGC demodulation system based on LABVIEW with sampling frequency 200 kHz and demodulates 400 Hz underwater acoustic signals. The experiment results show that this system can guarantee the interferometer to operate sensitively. The effectiveness and robustness of the proposed method is demonstrated via this experiment.

## Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C.J. Struck, Chair ASC S3
*CJS Labs, 57 States Street, San Francisco CA 94114-1401*

**Accredited Standards Committee S3 on Bioacoustics.** Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

*People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.*

**Scope of S3:** Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance and comfort.

WEDNESDAY MORNING, 4 DECEMBER 2013 UNION SQUARE 9, 10:45 A.M. TO 12:00 A.M.

## Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D.K. Delaney, Chair ASC S3/SC 1
*USA CERL, 2902 Newmark Drive, Champaign, IL 61822*

D.S. Houser, Vice Chair
*National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92016*

**Accredited Standards Committee S3/SC 1 on Animal Bioacoustics.** Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

*People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note - those meetings will be held in conjunction with the Standards Plenary meeting at 9:00 a.m. on Tuesday, 3 December 2013.*

**Scope of S3/SC 1:** Standards, specifications, methods of measurement and test, instrumentation and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria or aquariums; or free-ranging wild animals.

## Session 3pAA

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

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

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

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

**Chair's Introduction—1:00**

### *Invited Papers*

**1:05**

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

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

**2:05**

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

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

## Session 3pAB

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

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

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

### *Contributed Papers*

**1:00**

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

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

**1:15**

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

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

**1:30**

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

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



**1:45**

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

Studies with some echolocating odontocetes demonstrate that receiver-based automatic gain control (AGC) compensates for reductions in echo strength resulting from acoustic spreading loss. This study examined AGC in an echolocating bottlenose dolphin by measuring changes in hearing sensitivity over time courses corresponding to single click-echo pairs. The electrophysiological auditory steady-state response (ASSR) elicited by a 113-kHz sinusoidally amplitude-modulated tone was recorded while the dolphin performed a target discrimination task. Auditory electrophysiological responses were extracted from the instantaneous electroencephalogram and coherently averaged using the modulation rate of the 113-kHz tone as a reference. A Fourier transform was then performed with a 10-ms sliding window to obtain the ASSR amplitude as a function of time relative to the dolphin's outgoing click and received echo. The ASSR amplitude initially decreased at the time of click emission and then recovered over a course of 25 to 70 ms, depending on target range. This relatively long time course of

recovery appears to be consistent with forward-masking, as opposed to an AGC mechanism based on the contraction and gradual release of middle ear muscles coincident with click emission. [Work funded by SSC Pacific Naval Innovative Science and Engineering (NISE) program.]

**2:00**

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

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

**2:15**

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

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

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

# Session 3pBA

## Biomedical Acoustics: General Topics in Biomedical Acoustics

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

### *Contributed Papers*

**1:00**

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

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

**1:15**

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

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

1:30

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

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

1:45

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

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

2:00

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

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

2:15

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

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

2:30

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

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

2:45

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

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

3:00

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

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

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

# Session 3pEA

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

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

### *Contributed Papers*

1:30

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

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

1:45

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

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

**2:00**

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

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

**2:15**

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

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

**2:30**

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

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

## Session 3pED

## Education in Acoustics: Acoustics Education Prize Lecture

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

**Chair's Introduction—2:00**

### *Invited Paper*

**2:05**

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

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

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

## Session 3pID

## Interdisciplinary: Hot Topics in Acoustics

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

**Chair's Introduction—1:15**

### *Invited Papers*

**1:20**

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

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

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

**1:45**

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

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

**2:10**

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

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

**2:35**

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

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

## Session 3pNS

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

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

**Chair's Introduction—1:25**

### *Invited Papers*

**1:30**

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

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

**1:50**

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

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

**2:10**

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

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

**2:30**

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

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

## Session 3pPAa

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

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

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

***Invited Papers***

**1:00**

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

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



**1:20**

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

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

**1:40**

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

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

**2:00**

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

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

**2:20**

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

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

## *Contributed Papers*

**2:40**

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

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

**2:55**

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

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

# Session 3pPAb

## Physical Acoustics: Topics in Geophysical Acoustics

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

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

### *Contributed Papers*

**1:00**

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

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

**1:15**

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

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

**1:30**

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

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

**1:45**

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

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

3p WED. PM

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

**2:00**

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

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

**2:15**

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

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

**2:30**

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

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

**2:45**

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

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

## Session 3pSA

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

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

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

### *Invited Papers*

**1:00**

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

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

**1:20**

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

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

**1:40**

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

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

**2:00**

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

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

## *Contributed Paper*

**2:20**

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

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

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

# Session 3pSC

## Speech Communication: Communication Disorders

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

## *Contributed Papers*

**1:00**

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

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

**1:15**

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

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

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

**1:30**

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

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

**1:45**

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

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

**2:00**

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

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

**2:15**

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

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

## Session 3pSP

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

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

### *Contributed Papers*

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

## Plenary Session, Annual Meeting, and Awards Ceremony

James H. Miller, President
*Acoustical Society of America*

## Annual Meeting of the Acoustical Society of America

## Presentation of Certificates to New Fellows

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

Benjamin A. Cray – for contributions to underwater directional sensing

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

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

Tribikram Kundu – for contributions in nondestructive testing and evaluation

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

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

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

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

Kausik Sarkar – for contributions to the modeling of ultrasound microbubbles

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

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

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

## Award Announcements and Presentations of Awards

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

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

Rossing Prize in Acoustics Education to Juliette W. Ioup

Silver Medal in Biomedical Acoustics to Kullervo H. Hynynen

Silver Medal in Musical Acoustics to William J. Strong

## Session 3eED

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

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

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

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

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

## OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Wednesday are as follows:

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

# Session 4aAA

## Architectural Acoustics and Noise: Experience in Forensic Architectural Acoustics

Matthew V. Golden, Cochair
*Kinetics Noise Control, 6300 Irelan Place, Dublin, OH 43017*

David Lubman, Cochair
*DL Acoust., 14301 Middletown Lane, Westminster, CA 92683-4514*

### *Invited Papers*

**11:00**

**4aAA1. Developing criteria for identifying acoustical defects.** John LoVerde and David W. Dong (Veneklasen Associates, 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

In a construction defect lawsuit of a multifamily residential project, the determination of whether a defect exists often hinges on the criteria applied. For many acoustical items, such as plumbing and HVAC noise, there are no code requirements but a number of guidelines and recommendations. For items such as noise from traffic or airborne and impact sound isolation between units, minimum code requirements exist, but often a more stringent standard is applied. How does an expert decide when it is appropriate to apply an acoustical standard that is beyond that required by building codes? Project drawings, marketing materials, homeowner regulations, and other documents can provide indications of the intent and promise of the project as it relates to acoustical issues. The process is discussed with examples from recent cases.

**11:20**

**4aAA2. Construction defect actions in residential construction in California after passage of senate bill 800.** Pablo A. Daroux (Wilson, Ihrig & Associates, Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, pdaroux@wiai.com) and Terry Wilkens (Law Offices of Ann Rankin, Oakland, CA)

Construction defects actions (lawsuits) involving the sale of new residences that are sold on or after January 1, 2003, are governed by California Civil Code Section 986, also commonly referred to as “SB800” or the “Builder’s Right to Repair Law.” This legislation has redefined the rights and responsibilities of purchasers, sellers, builders, and designers regarding construction deficiencies in significant ways. This presentation will discuss the details of this Legislation as it affects the work of Consultants as Designers and as Forensic Experts.

### *Contributed Paper*

**11:40**

**4aAA3. Overcoming flooring impact isolation performance failures due to adjacent condominium ceiling deficiencies.** Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

There is an increasing demand to renovate apartments and condominiums by installing hard-surfaced flooring. It has become typical for a homeowner’s association (HOA) to enact regulations requiring that upper level dwelling units making flooring upgrades from soft-surface flooring to hard-surface flooring meet specific minimum field impact insulation class (FIIC) or impact sound rating (ISR) performance standards. It is common for these HOA requirements to apply only to the party making the flooring surface change. It is rare that the adjacent ceiling condition or ceiling renovation is addressed by such HOA requirements. This paper presents field performance test results for several renovated condominiums in different multi-family residential buildings that failed homeowner’s association impact performance requirements due to lower-level ceiling deficiencies. Examples of such failures are discussed and successful custom remedies are presented, some of which avoided threatened litigation.

# Session 4aAB

## Animal Bioacoustics and Noise: Bioacoustic Contributions to Soundscapes I

John Hildebrand, Cochair
*Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093*

Simone Baumann-Pickering, Cochair
*Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093*

### *Invited Papers*

**8:00**

**4aAB1. Bioacoustic contributions to the soundscape.** John Hildebrand and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu)

The soundscape, the combination of all sounds found in the environment, has been divided into three components: geophony—sounds that are generated by non-biological sources such as wind and waves; biophony—sound generated by animals, excepting humans; and anthrophony—sound generated by human activities. The soundscape can be further divided by spatial, temporal, and frequency band variations. Spatial variations occur both owing to proximity to various sound sources and to the sound propagation environment. Temporal variations occur on seasonal, lunar, daily, and other cycles. Use of different frequency bands may be a response to spatial and temporal overlap of sounds. In this session, we consider biological components of the soundscape and how sounds produced by animals interact with each other and with non-biological and anthropogenic sounds. We suggest that studies using long-term passive acoustic monitoring may take a more holistic approach to analysis using the concept of soundscape. This approach emphasizes the connections and relations between sound events, rather than focusing on the extraction of individual sound events. The session draws on data collected both in terrestrial and marine settings to illustrate how the soundscape concept can be applied to better understand animal use of sound.

**8:20**

**4aAB2. Soundscape ecology: A review of a new synthesis area of acoustics of landscapes.** Bryan C. Pijanowski and Luis Villanueva-Rivera (Forestry and Natural Resources, Purdue Univ., 195 Marsteller St., 305 FORS Bldg., West Lafayette, IN 47906, bpijanow@purdue.edu)

Soundscape ecology is an emergent area of acoustics that attempts to synthesize the concepts of landscape ecology, bioacoustics, noise, music, ethics, and biogeography. By focusing on the interplay of three main sources of sound: biological, geophysical, and anthropogenic, we hope to understand how humans impact ecosystems at a variety of spatial and temporal scales. Another focus of our work is the identification of special ecological places that possess unique and highly valued soundscapes. I will review the current state of the science and efforts to move this field forward using the expertise across these varied disciplines along with the work that we are conducting in the temperate forest, tropical, and desert ecosystems of North America.

**8:40**

**4aAB3. Composing soundscapes from real-time acoustic data streams.** Michel Andre (Lab. of Appl. BioAcoust., Tech. Univ. of Catalonia, BarcelonaTech, UPC, Rambla Exposicio 24, Vilanova i la Geltrú, Barcelona 08800, Spain, michel.andre@upc.edu) and M. Andre (Ocean Sound Sci., CSA Ocean Sci. Inc., Stuart, FL)

The growing scientific and societal concern about the effects of underwater sound on marine ecosystems has been recently recognized through the introduction of several international initiatives aiming at measuring the environmental impact of ocean noise on large spatial and temporal scales. From a regulatory perspective, the European Marine Strategy Framework Directive includes noise as one of 11 descriptors to determine Good Environmental Status of the oceans. The Directive specifically requires Member States to provide a measure of annually averaged noise. The LAB has developed a software package that measures sound levels and monitors acoustic sources in real-time; this software was used for the LIDO project and is now operating to provide industry with an environmentally responsible approach (CSA, www.oceansound.com). The system is currently operating worldwide from several wired and radio-linked observatories. Recently, through a zero-cost contract with the CTBTO (Preparatory Commission for the Comprehensive nuclear-Test Ban Treaty Organization), years of data from hydroacoustic stations were analyzed to look for background noise trends and to detect cetacean presence. Here, we present the analysis of four CTBTO platforms, each covering 42 months of data, focusing especially on the estimation of background noise levels and the measurement of noise contributions from anthropogenic sources.

**9:00**

**4aAB4. From pole to pole: Soundscapes of the Atlantic Ocean.** Holger Klinck (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Hatfield Marine Sci. Ctr., 2030 SE Marine Sci. Dr., Newport, OR 97365, Holger.Klinck@oregonstate.edu), Jennifer L. Miksis-Olds (Appl. Res. Lab., Penn State Univ., State College, PA), Sharon L. Nieukirk, Haruyoshi Matsumoto, and Robert P. Dziak (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR)

Between July 2009 and December 2010, two identical calibrated hydrophone packages were deployed and continuously operated (2,000 Hz sampling rate) in the Fram Strait (79°N, 5.5°E) and the Bransfield Strait (62°S, 55.5°W). Analysis of these recordings were combined with a data set collected during the same time period by the Comprehensive Nuclear-Test-Ban Treaty Organization International Monitoring System (CTBTO IMS) hydroacoustic station HA10 (250 Hz sampling rate) at Ascension Island (8°S, 14.4°W). The combination of these datasets allowed a comparison of low-frequency noise levels in polar and tropic areas of the Atlantic Ocean. The recordings were analyzed for major natural (e.g., marine mammals, ice) and anthropogenic (e.g., shipping, seismic) contributors to the ambient sound field and their seasonal variability. Preliminary results indicate (1) a higher seasonal variability of ambient noise levels in polar regions compared to the tropics, (2) the seasonal variability of ambient noise levels in the Arctic and Antarctic were driven by different contributors, and (3) highest noise levels were observed in the Arctic in association with seismic oil and gas exploration. [Partial funding from NOAA/PMEL and the Korea Polar Research Institute.]

**9:15**

**4aAB5. Nonlinear time series analysis of snapping shrimp sounds.** Tyler Hee Wai, John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), John Gebbie, and Martin Siderius (Elec. Eng., Portland State Univ., Portland, OR)

Snapping shrimp produce sounds through cavitation bubbles generated by the rapid closing of their claws. These sounds are a primary source of ambient noise in sub-tropical, shallow water environments. Though seasonal and daily variations of snapping shrimp sound levels have been reported, comprehensive studies of short time series have been limited. We investigate the respective spectral characteristics from acoustic arrays recordings at various locations in Oahu, HI, over hourly, daily and weekly periods. Nonlinear time series analysis methods encompassing recurrence plots are used to investigate the acoustic time series together with bivariate and cross recurrence plot analysis with that of tidal and sunlight cycles. Signal entropy and determinism are investigated and transient acoustic ship sounds are located

**9:30**

**4aAB6. The soundscape experienced by the Southern White rhinoceros (*Ceratotherium simum simum*) at a wildlife park conservation center.** Suzi Wiseman (Geography, Texas State Univ.-San Marcos, 601 University Dr., San Marcos, TX 78666, sw1210txstate@gmail.com) and Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Many creatures, including the myopic rhinoceros, depend on hearing and smell even more than on their sight. Noise impacts human health and reproduction, and may impact these other mammals similarly, or even more. Rhinos have been recorded vocalizing infrasonically and sonically. They have a poor breeding record in urban zoos, in which infrasonic noise tends to be chronic. Herd size and composition, the age of potential mates, substrate, enclosure design, and other factors have been studied but little attention if any has been paid to soundscape. As a first step to comparing the soundscapes of facilities in which rhinos have and have not bred successfully, this project recorded and analyzed the soundscape of white rhinos at Fossil Rim Wildlife Center in Texas, one of five Conservation Centers for Species Survival, and one of the few U.S. facilities to successfully breed this species in recent years. Animal responses to sound have been shown to depend on sound level, rate of onset, duration, number of events, spectral distribution of the sound energy, presence of pure tones, the relative level of background noise, and semantics. Similar parameters were analyzed for the data recorded at Fossil Rim and will be presented here.

**9:45**

**4aAB7. An automatic single station multipath ranging technique for 20 Hz fin whale calls: Applications for ocean bottom seismometer data in the northeast Pacific.** Michelle Weirathmueller and William S. D. Wilcock (Univ. of Washington, 1503 NE Boat St., Seattle, WA 98105, michw@uw.edu)

Ocean bottom seismometers (OBSs) deployed to monitor earthquakes also record 20 Hz fin whale vocalizations. Because OBSs are often deployed too far apart to record a call on more than one instrument, and call detections are dependent on the acoustic environment, techniques are required to determine call density from a single instrument. One approach is to use point transect distance sampling, which relies on estimates of horizontal range to each call. An automated technique has been developed to estimate range based on the amplitude and relative timing of multipath arrivals assuming a uniform water depth. The method has been evaluated using independently located calls recorded by a seismic network on the Endeavour Segment of the Juan de Fuca Ridge, a region of rough and partially sedimented seafloor. Results suggest that the method works well to ranges of ~5 km but at larger ranges ambiguities arise. Calls at >10 km range are sometimes located between 5 and 10 km because the times are erroneously fit by one too few water column multiples and the amplitudes have too much scatter to be diagnostic of range. The technique is presently being applied in different acoustic settings to determine whether it is more reliable in flat sedimented regions.



**10:00–10:15 Break**

*Invited Papers*

**10:15**

**4aAB8. Comparison of soundscapes across the Bering Sea shelf, a biological perspective.** Jennifer L. Miksis-Olds (Appl. Res. Lab, Penn State, PO Box 30, M.S. 3510D, State College, PA 16804, jlm91@psu.edu)

Selectively decomposing and visualizing different aspects of an acoustic time series provides a greater understanding of the sources and environmental dynamics contributing to and shaping the temporal and spatial patterns of the measured sound. Source contributions to the overall soundscape vary in space and time, and the biological component is highly dependent on temporal, geographic, and oceanographic factors. The biological contributions to soundscapes along the eastern Bering Sea shelf are tightly coupled to the presence of sea ice and have strong annual patterns. The differences between locations reflect the relative contribution of different species to the

soundscape. Using soundscape plots to display the acoustic environment during the overlap of peak vocal activity by different marine mammal species in the winter/spring season produces a visual representation of acoustic niche partitioning by Arctic species.

**10:35**

**4aAB9. Larval settlement in response to estuarine soundscapes.** David Eggleston, Ashlee Lillis, and Del Bohnenstiehl (Dept. of Marine, Earth & Atmospheric Sci., NC State Univ., Raleigh, NC 27695-8208, eggleston@ncsu.edu)

Ambient underwater sound has the potential to be an important orientation and settlement cue for marine invertebrate larvae, yet larval responses to relevant sound patterns are largely unknown. In estuaries of the Southeastern United States, oyster reefs are patchy productive habitats that harbor many soniferous fish and invertebrates, creating distinct sound characteristics. This habitat-related sound could provide a useful cue for the planktonic larvae of obligate reef dwellers and facilitate encounter with suitable settlement substrate. To investigate sound as a settlement cue in this system, larval settlement responses to oyster reef and soft-bottom sounds, as well as a no-sound control were tested for the Eastern oyster, *Crassostrea virginica*. Laboratory and field experiments suggest that sound has a significant effect on oyster settlement rates: higher numbers of larvae settled in the presence of oyster reef sounds than in soft-bottom sound or silent control treatments. Improved understanding of the relationship between habitat sound fields and subsequent larval recruitment is central to bio-physical studies of larval connectivity and recruitment in marine systems.

## *Contributed Papers*

**10:55**

**4aAB10. Locating invertebrate sound sources, including hermit crabs, on shallow water reefs in the Northwestern Hawaiian Islands using an L-shaped array of hydrophones.** Simon E. Freeman, Lauren A. Freeman (Scripps Inst. of Oceanogr., 3725 Miramar St. Apt. C, La Jolla, CA 92037, sfreeman@ucsd.edu), Marc O. Lammers (Oceanwide Sci. Inst., Honolulu, HI), and Michael J. Buckingham (Scripps Inst. of Oceanogr., La Jolla, CA)

Using ambient noise for extracting ecological information from coastal waters is a tantalizing idea that has gained momentum with the increasing use of long-term passive recorders. Single-element hydrophone recordings from different reef locations reveal substantial variation in the biologically produced sound field, the spatial scales and sources of which are poorly understood. A seven-element L-shaped array was deployed in a spur-and-groove coral reef environment at Kure Atoll in the Papahanaumokuakea Marine National Monument, Northwestern Hawaiian Islands, in an effort to resolve small-scale spatial variability in reef-generated ambient noise. For each given time step, a fast, pair-wise coherence technique was initially used to estimate correlation in the ambient noise field. Curved-wavefront adaptive beamforming was then performed within range and azimuth windows that encompassed each estimated source location (areas of high correlation). Array processing performance was evaluated and frequency-dependent, spatiotemporal distribution of the sound field derived. The environment surrounding the array was photographically surveyed and ecologically classified through SCUBA-based observations. Survey data were used in an attempt to identify reef structures and the organisms that were the dominant contributors to the local sound field, notably hermit crabs and other hard-shelled invertebrates.

**11:10**

**4aAB11. A disproportion in bowhead whale call counts around an acoustic array during the fall 2012 migration in the Chukchi Sea, Alaska: An investigation into the potential of acoustic masking.** Jennifer L. Wladichuk (JASCO Appl. Sci., 2305 – 4464 Markham St., Victoria, BC V8Z 7X8, Canada, jennifer.wladichuk@jasco.com), Julien Delarue, Bruce Martin (JASCO Appl. Sci., Dartmouth, NS, Canada), Xavier Mouy, and Heloise Frouin-Mouy (JASCO Appl. Sci., Victoria, BC, Canada)

A long-term passive acoustic study in the Chukchi Sea monitors the bowhead whale (*Balaena mysticetus*) migration between their summering and wintering grounds in the Beaufort and Bering Seas, respectively. A hexagonal array offshore from Point Lay, AK was located near the center of their migratory corridor in the falls of 2009 through 2011. In 2012 however, there was a large difference in the number of acoustic detections across the array, with call counts close to three times higher in its northwest section compared to its southeast. An increase in vessel traffic in the area resulted in higher ambient noise conditions for much of the summer. This paper investigates the possibility of acoustic masking of calls due to this increase in background noise levels.

**11:25**

**4aAB12. Passive acoustic monitoring of biological and anthropogenic sounds at America's first offshore wind farm.** T. Aran Mooney, Maxwell B. Kaplan, Luca Lamoni (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS50, Woods Hole, MA 02543, mkaplan@whoi.edu), Aimee Boucher (Nicholas School of the Environ., Duke Univ., Durham, NC), and Laela S. Sayigh (Biology Dept., Woods Hole Oceanogr. Inst., Woods Hole, MA)

Cape Wind, situated in Nantucket Sound, Massachusetts, is poised to become America's first offshore windfarm. Our objective is to establish baseline (pre-construction) sound levels of human and biological activity, including diel and seasonal variability of various sound types, at the construction site and three nearby comparison sites. Acoustic recorders have been deployed since April 2012, recording on a 10% duty cycle (sample rate: 80 kHz). Biological contributions to the local soundscape are primarily fish sounds, with the dominant signal likely being cusk eel (Family Ophidiidae) calls. These calls, which are composed of stereotyped pulses with an average bout duration of 3.3 $\pm$0.8 s and mean peak frequency of 1030 $\pm$200 Hz, show both seasonal and diel variation. Dense choruses were detected during summer (July), but limited activity occurred in the fall and winter. During vocal periods, detections occurred throughout the day but peaked near dusk. Vessel traffic also showed diel and seasonal trends, with peaks during the daytime and in the summer, which indicates that boat activity can be tracked acoustically. These trends in biological and anthropogenic activity provide key baseline records for evaluating the influence of windfarm construction and operation on a local US soundscape.

# Session 4aAO

## Acoustical Oceanography and Animal Bioacoustics: Properties, Trends, and Utilization of Ocean Noise I

Jennifer L. Miksis-Olds, Cochair
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Zoi-Heleni Michalopoulou, Cochair
*Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102*

**Chair's Introduction—7:55**

### *Invited Papers*

**8:00**

**4aAO1. Ocean sensing using ambient noise: An overview.** Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

The random nature of ocean ambient noise, natural or man-made, tends to suggest limited utility: ambient noise is commonly considered to be a nuisance for acoustic sensing or imaging applications. However, whether noise is a nuisance or a useful signal actually depends on how it is processed. Indeed, acoustic fields from random sources are often considered to be incoherent, but there is some coherence between two sensors that receive signals from the same individual noise source. Consequently, by cross-correlating the ambient noise recorded at two hydrophones one can estimate the acoustic waves that propagate between them, thus yielding an estimate of the Green's function between those sensors. This passive approach provides a foundation for ocean sensing and imaging techniques using only ambient noise, thus without active transmitters. We will examine the basic background physics of extracting coherent information from ambient noise. Specifically we will highlight among others the role of frequency/bandwidth, noise structure/distribution, ocean fluctuations, sensor separation and, if employed, the array configuration on this process. Further we will give an overview of recent experimental results in shallow and deep water and discuss practical challenges for implementing noise-based ocean sensing techniques.

**8:45**

**4aAO2. Extracting seabed properties using ocean noise measured on an autonomous underwater vehicle mounted array.** Martin Siderius (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Portland, OR 97201, siderius@pdx.edu), Peter L. Nielsen (STO-CMRE, La Spezia, Italy), and James H. Miller (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

Ocean ambient noise has been used to estimate seabed properties such as bottom loss and bottom layering. These ambient noise methods use beamforming and typically collect data on hydrophone arrays of 16 to 32 sensors that are moored or allowed to drift. Compact arrays containing only a few sensors can be mounted to autonomous underwater vehicles (AUVs). This configuration provides better control over the location of the measurement compared to a drifting system. However, compact arrays of this type present a challenge due to limited beamforming capabilities. In July 2012 and June 2013, the Centre for Marine Research and Experimentation conducted the GLASS'12 and GLASS'13 experiments to investigate the possibility of using an AUV equipped with a compact nose array for seabed characterization. The nose-mounted array consisted of a 5-hydrophone vertical and 4-hydrophone tetrahedral array. In this paper, the results from these experiments will be presented along with approaches to seabed characterization using hydrophone arrays with few sensors.



**9:05**

**4aAO3. On the feasibility of estimating attenuation from the coherence of ambient noise.** Ravi Menon and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., MC-0238, UC San Diego, La Jolla, CA 92093, rmenon@ucsd.edu)

There has been a long standing interest, especially in seismology, to be able to reliably retrieve attenuation information from ambient noise. While theoretical work has shown that it is possible to retrieve attenuation from noise, in practice, several factors such as anisotropy of noise distribution and intensity, site amplification factors, etc., make it an extremely challenging task. By assuming a sufficiently uniform noise distribution, researchers have obtained estimates of attenuation by fitting the theoretical noise coherence function modified by an exponentially decaying term, to the observed coherence. In this talk, we present results from analyzing seismic data from the Southern California seismic network in the microseism band (0.05–0.2 Hz) and discuss the feasibility of estimating attenuation from coherence. Specifically, we demonstrate that the assumption of a single wavespeed at each frequency is not valid at certain frequencies. Such variations in wavespeed are often due to inhomogenities in the medium and scattering effects, and it is very likely that interactions between these waves results in a reduction in amplitude due to phase cancellations, which might be erroneously perceived as attenuation. While it might still be possible to estimate attenuation if these interaction effects are accounted for, it raises questions on the validity of the simple exponential decay model.

**9:25**

**4aAO4. Offshore impact pile driving as a source of opportunity for geoacoustic investigations.** Gopu R. Potty, James H. Miller, and Huikwan Kim (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, potty@egr.uri.edu)

Noise generated by offshore impact pile driving can radiate into and propagate through the air, water, and sediment. Most of the recent studies have focused on predicting acoustic pressure field in water to assess environmental impacts. In this study, we focus on the propagation of acoustic energy along the interface between water and ocean bottom. We modeled the interface wave propagation using the commercial Finite Element (FE) code, ABAQUS 6.11. A field test is planned for this summer using a scaled model impact pile driving off the dock in the Bay Campus University of Rhode Island (URI). Interface data (particle velocities at the water-sediment interface) will be collected using the Shear Measurement System. Our efforts will focus on identifying the arrivals corresponding to various wave types using data-model comparison. In addition to the interface waves, we will also model the Mach wave front arrival angle. We will explore the possibility of utilizing the arrival times and angles corresponding to these wave types for setting up an inverse problem. This test will also verify the possibility calculating compressional wave speed using a single three-axis geophone by measuring arrival angle of Mach wave front at the interface generated by impact pile driving. [Work sponsored by the Link Foundation Ocean Engineering and Instrumentation PhD Fellowship program.]

**9:45**

**4aAO5. Geoacoustic inversion of noise radiated by surface ships.** David P. Knobles, Robert A. Koch, Jason D. Sagers, and Steven A. Stotts (ARL, UT at Austin, 10000 Burnet Rd., Austin, TX 78758, knobles@arlut.utexas.edu)

Addressed is the issue of determining physical properties of the seabed in shallow water using ambient noise from surface ships of opportunity. Both the amplitude and phase of the source spectrum are unknown. The coherent modal interference structure in the received spectra is sensitive to parameters such as the sound speed in the seabed, the speed of the ship, the source-receiver range, the water depth, etc. Such sensitivity allows in some cases for information on these parameters to be extracted from a statistical information approach such as maximum entropy. In shallow water, the rate of decay with propagation range of the received incoherent levels is sensitive to the intrinsic seabed attenuation. For ship of opportunity data, estimating values for the attenuation is complicated by the ambiguity existing between source level and attenuation. This work explores various methods that attempt to resolve the source level and attenuation ambiguity using experimental data taken in the Gulf of Mexico and the New Jersey continental shelf. One approach is to use short-range data to give a prior distribution for the source levels and to then estimate the marginal distributions of the attenuation with longer-range data. [Work supported by ONR.]

**10:05–10:20 Break**

## *Contributed Paper*

**10:20**

**4aAO6. Radiated ship noise level estimates from measurements in a fjord.** Stian Coward (Norwegian Univ. of Sci. and Technol., Trondheim, Norway), Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no), and Hefeng Dong (Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

This paper presents estimates of radiated ship noise levels (20 Hz—1 kHz) due to commercial shipping in the Oslofjord (Norway). Data were recorded at two Networked Intelligent Underwater Sensors (NILUS) units, each equipped with an omni-directional hydrophone placed near the seabed, at water depths of 45 and 110 m, respectively. Received noise levels due to traffic in two nearby shipping lanes (ranges, 0.4–3.0 km) were for these non-ideal test site conditions processed and corrected to monopole source levels by use of the RAM propagation model with range estimates from auxiliary data, environmental input from a prior survey of the area, and a vertically distributed source model [Trevorrow *et al.*, J. Acoust. Soc. Am. **124**(2), 767–778] to model uncertain source depth. Estimates of source spectra and of broadband source levels of cargo ships are presented and compared with deep-water measurements from the literature.

## *Invited Papers*

**10:35**

**4aAO7. Models for non-stationary reverberation noise.** Leon Cohen (Physics-Hunter, City Univ. of New York, 695 Park Ave., New York, NY 10065, leon.cohen@hunter.cuny.edu)

The random phasor approach to reverberation noise gives Gaussian statistics for intensity and is stationary in time and space. We show that fundamentally different things happen when the noise is defined by a superposition of random propagating pulses that have width. We calculate the nonstationary autocorrelation function of such a model and describe how it evolves in time and space in a dispersive medium. Furthermore, we give criteria for when a nonstationary autocorrelation function is locally stationary in time and/or space. We also consider how and under what conditions the noise evolves to the stationary case given by the ransom phasor sum. [Work supported by ONR.]

10:55

**4aAO8. The effect of seawater attenuation on the directionality and spatial coherence of surface-generated ambient noise.**
Michael J. Buckingham (SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

Acoustic attenuation in seawater is sufficiently weak that it usually has little effect on the spatial characteristics of ambient noise. However, at frequencies above 10 kHz and depths below 6 km, seawater attenuation may influence the directionality of surface-generated ambient noise. In a semi-infinite, homogeneous ocean, all the noise is downward-traveling, since there are no bottom reflections, and, in the absence of attenuation, the directional density function varies as the cosine of the polar angle measured from the zenith. When attenuation is present, the angular width of this noise lobe becomes narrower, because sound from distant surface sources is attenuated more than acoustic arrivals from overhead. This narrowing effect increases as frequency rises, since the attenuation is a rapidly increasing function of frequency. The spatial coherence of the noise is also modified by the attenuation. For a pair of horizontally (vertically) aligned sensors, the zeroes in the spatial coherence function are shifted to higher (lower) frequencies. These effects of attenuation on the noise field are sufficiently large to be detectable using the recently developed, deep diving instrument platform Deep Sound, which is capable of measuring broadband ambient noise on multiple hydrophones to depths of 11 km. [Research supported by ONR.]

11:15

**4aAO9. Ambient noise modeling using sound field reciprocity.** David R. Barclay and Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., M.S. #11, Woods Hole, MA 02543, dbarclay@whoi.edu)

An efficient numerical calculation of the vertical and horizontal coherence of the noise field in the ocean due to a distributed sheet of sources can be carried out using a parabolic equation (PE) propagation model and the reciprocity property of the complex pressure field. In the case of an infinite sheet of sources near the surface in an infinitely deep, isovelocity ocean, the calculation of vertical coherence using this method is in agreement with the Cron and Sherman model for wind-driven surface noise, as well as with measurements made of vertical coherence in the Philippine Sea. This numerical model also gives the effective surface-listening radius for a single hydrophone by calculating the contribution to the noise field at the receiver from a small noise patch at the surface, as a function of range. The model is capable of calculating the vertical and horizontal coherence (directionality) of the noise field and the depth-dependent surface-listening radius of a hydrophone in a shallow water waveguide and over range-dependent environments, such as a wedge, canyon, or shelf break. Additionally, this technique can be used to calculate the spatial properties of the noise field due a finite patch of surface sources in motion, such as a rainstorm.

11:35

**4aAO10. Computation of ocean noise fields.** Michael B. Porter and Laurel J. Henderson (HLS Res., 3366 N. Torrey Pines Ct., Ste 310, La Jolla, CA 92037, mikeporter@hlsresearch.com)

Ships and winds provide two key examples of sound sources that “illuminate” the ocean. In recent years, we have seen a great interest in such noise fields. It has become clear that this noise field provides an important signal that can be used to image the environment and the ocean bottom in particular. Separately, conservationists have become very interested in the fields in terms of their impact on marine life. The modeling of such noise fields provides interesting challenges. First, the noise field is the result of not just a single propagation calculation but a fan of calculations connecting a receiver location to all the noise sources. Second, the characteristic cylindrical spreading from a point source is counter-balanced nearly perfectly by the increase in area of a cylindrical annulus as we accumulate the contributions of noise sources in range. Thus, the noise can travel to extremely long range. This talk will discuss such noise calculations in terms of both the numerics and the physics. We will discuss numerical options in terms of efficiency and accuracy. In addition, we will discuss the role of environmental uncertainties.

## Session 4aBA

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

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

Gail ter Haar, Cochair
*Phys., Inst. of Cancer Res., Royal Marsden Hospital, Downs Rd., Sutton SM25PT, United Kingdom*

**Chair's Introduction—7:55**

### *Invited Papers*

**8:00**

**4aBA1. Challenges in the characterization of high intensity therapeutic ultrasound devices and fields and regulatory guidance development.** Subha Maruvada and Gerald R. Harris (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993, subha.maruvada@fda.hhs.gov)

As part of its regulatory responsibility for medical devices, the US Food and Drug Administration (FDA), Center for Devices and Radiological Health, reviews pre-clinical and, if necessary, clinical data submitted by device applicants relevant to safety, effectiveness, and accurate labeling. For HITU devices, pre-clinical testing can include ultrasound power measurement, pressure/intensity mapping, temperature field characterization (*in vitro* and *in vivo*), acoustic and thermal simulations, and demonstration of the targeting accuracy and treatment monitoring. To assist manufacturers and third parties in the submission process, the FDA provides pre-market submission guidance that often relies on technical standards and specifications such as published by the International Electrotechnical Commission (IEC). IEC documents for HITU are currently nearing completion. However, even with these documents, technical challenges remain because of the focused, large amplitude pressure fields encountered in HITU. Measurement and modeling issues include using radiation force balances at HITU power levels, the need for complex deconvolution with some hydrophone types, validation of simulation models, tissue-mimicking material development for temperature measurements, and cavitation detection and quantification. These challenges along with current FDA guidance efforts will be discussed.

**8:20**

**4aBA2. Standards development and execution for therapeutic ultrasound applications.** Mark E. Schafer (Sonic Tech, Inc., 23 Brookline Court, Ambler, PA 19002-1904, marks@sonictech.com)

Therapeutic ultrasound holds great promise in the treatment of a number of different disease states and patient conditions. In bringing these devices to market, the two main considerations are naturally safety and efficacy. Central to the safety evaluation of such devices is the determination of potential hazards through the use of international standards. The International Electrotechnical Commission (IEC), has several standards and initiatives in the area of therapeutic ultrasound, both in the measurement of ultrasonic fields and in safety characterization. To be effective, these standards must not only be technically accurate, but also sufficiently practical that their primary users, namely manufacturers, can properly implement them. This talk reviews the development of standards related to therapeutic ultrasound and also discusses the implementation approaches among different manufacturers. For instance, while diagnostic ultrasound systems can be safely managed with only type-testing, the energy levels from therapeutic systems require that every system be evaluated or calibrated before patient use. Typically, systems undergo detailed sub-system testing, with only limited testing performed on the final product at full energy levels. Assumptions with regard to output linearity are also used to further minimize the difficulties implicit in measurements at high ultrasonic energies.

**8:40**

**4aBA3. Just doing our DUTy!—An international project on dosimetry for ultrasound therapy.** Adam Shaw (Acoust. and Ionizing Radiation Div., National Physical Lab., Hampton Rd., Teddington, Middlesex TW11 0LW, United Kingdom, adam.shaw@npl.co.uk)

This presentation will describe a large international project which aims to develop the metrological infrastructure for the determination of ultrasound exposure and dose to tissue. Ultimately this will improve treatment planning and risk assessment. The 3-year project started in June 2012. It is co-ordinated by the UK National Physical Laboratory and is funded in part by the European Metrology Research Programme of the European Union. The other project partners are INRIM (IT), PTB (DE), UME (TR), CSIC (ES), ICR (UK), Moscow State University (RU), NIM (CN), and University of Merseburg (DE). The scientific work is divided into six Work Packages: —Quantities and definitions will review dose and *in situ* exposure quantities proposing alternatives where necessary; —Laboratory dosimetry standards will develop reference methods for thermal and non-thermal therapeutic dose parameters; —Dose modeling and

validation will develop theoretical and computer modelling procedures to calculate acoustic and thermal dose quantities with well understood uncertainties; —Intercomparison of methods will formally compare measurement methods and models developed; —Dosimetry Transfer Standards will develop phantoms and test systems for the assessment of clinical ultrasound equipment, transferring laboratory standards to end-users; —Application to clinical treatment will provide a direct link between the methods developed and the clinical use of therapeutic ultrasound with the long-term aim of improving treatment planning.

**9:00**

**4aBA4. Dosimetry for therapy ultrasound.** Gail ter Haar (Phys., Inst. of Cancer Res., Royal Marsden Hospital, Downs Rd., Sutton SM25PT, United Kingdom, gail,terhaar@icr.ac.uk)

Ultrasound is probably the only clinical therapy modality that does not as yet have a well established dosimetric parameter. There are a number of reasons for this. Unlike other energy forms, ultrasound produces cell killing by two distinct principle mechanisms: cavitation and heating. While dose parameters have been proposed for each mechanism, there has been no attempt to combine these into a single unit. Existing dosimetric methods will be reviewed and put into the context of other therapeutic methods. Biological and physical end points will be discussed.

**9:20**

**4aBA5. Addressing nonlinear propagation effects in characterization of high intensity focused ultrasound fields and prediction of thermal and mechanical bioeffects in tissue.** Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, vera@apl.washington.edu), Petr V. Yuldashev (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Wayne Kreider, Oleg A. Sapozhnikov, Michael R. Bailey, Tatiana D. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), Adam D. Maxwell (Dept. of Urology, Univ. of Washington, Seattle, WA), and Lawrence A. Crum (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

Nonlinear propagation effects are present in most fields generated by high intensity focused ultrasound (HIFU) sources. In some newer HIFU applications, these effects are strong enough to result in the formation of high amplitude shocks that actually determine the therapy and provide a means for imaging. However, there is no standard approach yet accepted to address these effects. Here, a set of combined measurement and modeling methods to characterize nonlinear HIFU fields in water and predict acoustic pressures in tissue is presented. A characterization method includes linear acoustic holography measurements to set a boundary condition to the model and nonlinear acoustic simulations in water for increasing pressure levels at the source. A derating method to determine nonlinear focal fields with shocks in situ is based on the scaling of the source pressure for data obtained in water to compensate for attenuation losses in tissue. The accuracy of the methods is verified by comparing the results with hydrophone and time-to-boil measurements. Major effects associated with the formation of shocks are overviewed. A set of metrics for determining thermal and mechanical bioeffects is introduced and application of the proposed tools to strongly nonlinear HIFU applications is discussed. [Work supported by NIH EB007643, T32 DK007779, and RFBR 13-02-00183.]

## *Contributed Papers*

**9:40**

**4aBA6. Holography and numerical projection methods for characterizing the three-dimensional acoustic fields of arrays in continuous-wave and transient regimes.** Wayne Kreider (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Adam D. Maxwell (Dept. of Urology, School of Medicine, Univ. of Washington, Seattle, WA), Petr V. Yuldashev (Faculty of Phys., Moscow State Univ., Moscow, Russian Federation), Bryan W. Cunitz, Barbrina Dunmire (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov, and Vera A. Khokhlova (Faculty of Phys., Moscow State Univ., Moscow, Russian Federation)

The use of projection methods is increasingly accepted as a standard way of characterizing the 3D fields generated by medical ultrasound sources. When combined with hydrophone measurements of pressure amplitude and phase over a surface transverse to the wave propagation, numerical projection can be used to reconstruct 3D fields that account for operational details and imperfections of the source. Here, we use holography measurements to characterize the fields generated by two array transducers with different geometries and modes of operation. First, a seven-element, high-power therapy transducer is characterized in the continuous-wave regime using holography measurements and nonlinear forward-projection calculations. Second, a C5-2 imaging probe (Philips Healthcare) with 128 elements is characterized in the transient regime using holography measurements and linear projection calculations. Results from the numerical projections for both sources are compared with independent hydrophone measurements of select waveforms, including shocked focal waveforms for the therapy transducer. Accurate 3D field representations have been confirmed, though a notable sensitivity to hydrophone calibrations is revealed. Uncertainties associated with this approach are discussed toward the development of holography measurements combined with numerical projections as a standard metrological tool. [Work supported by NIH EB007643, P01DK043881, R01DK092197 and T32DK007779, NSBRI through NASA NCC 9-58, and RFBR 13-02-00183.]

**9:55**

**4aBA7. High pressure focused ultrasound field characterization.** Baki Karaböce (Ultrasound Lab., TUBİTAK UME, Beşevler, Gebze, Kocaeli 41470, Turkey, baki.karaboce@tubitak.gov.tr), Ali Şahin (Phys. Dept., İnönü Univ., Malatya, Turkey), Eyüp Bilgiç, Enver Sadıkoğlu (Ultrasound Lab., TUBİTAK UME, Gebze, Turkey), and Süreyya Nur (SHMYO, İnönü Univ., Malatya, Turkey)

High intensity focused ultrasound (HIFU) transducers are novel and very attractive tools for cancer therapy. They produce very high pressure acoustic fields up to tens of MPa at the focus; thus, acoustic characterization of HIFU fields must be investigated in order to ensure the safe and effective use in clinical applications. A needle hydrophone has been used for HIFU transducer characterization in the newly designed home made system at TÜBİTAK UME (The Scientific and Technological Research Council of Turkey, the National Metrology Institute) Ultrasound laboratory. TÜBİTAK UME HIFU measurement system was controlled by a LABVIEW based data translation program. The driving signal was generated by signal generation card and output from the hydrophone was fed directly into a DSO oscilloscope card. The controlling program executed a "capture-analyze-move" cycle, allowing a large number of measurements to be made. Field scanning measurements are made between 1 MPa, 2 MPa, and 3 MPa pressures. Theoretical model for the beams of periodic waves with an initially uniform amplitude distribution was also performed, based on the Khokhlov-Zabolotskaya-Kuznetsov equation. Numerical solutions were compared with the experimental data and found to be in agreement.

**10:10–10:30 Break**

**10:30**

**4aBA8. Feasibility of using infra-red system for absolute calibration of high intensity focused ultrasound fields.** Svetlana Shmeleva (Dept. of Medical Phys., Moscow State Univ., Leninskie gori 1/2, Moscow 119991, Russian Federation, sveta@acs366.phys.msu.ru), Vera Khokhlova (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), Leonid Gavrilov (N.N. Andreyev Acoust. Inst., Moscow, Russian Federation), Eleanor Martin, Neelaksh Sadhoo, and Adam Shaw (Acoust. and Ionising Radiation Div., National Physical Lab., Teddington, United Kingdom)

In recent years, considerable progress has been achieved in the use of infrared (IR) method for measuring acoustic fields of high intensity focused ultrasound (HIFU) transducers. The principle of the method is to determine the intensity distribution at the surface of an acoustic absorber by measuring the distribution of the initial rate of temperature increase at the start of insonation. Here, the method is extended to estimate the absolute values of intensity in ultrasound fields of HIFU transducers. The approach compares the temperature rise measured at the surface of a thin absorber using an IR camera and the pressure distribution measured in water using a hydrophone. The measurements were carried out for two focused HIFU transducers and a flat physiotherapy transducer. During the IR measurement, the transducer was driven in tone burst mode to avoid interference from reflected acoustic waves between the absorber/air interface and the transducer. The method enables fast quantitative estimations of intensity distributions that agree well with hydrophone measurements and gives reasonably consistent results for medical transducers of different geometries. [Work was supported in parts by RFBR 12-02-31388, 12-02-00028, 13-02-00183; NIH 2R01EB007643-05, the European Metrology Research Programme (Joint Research Project HLT03) and the UK National Measurement Office.]

**10:45**

**4aBA9. Sensitivity and power handling capacity of the Ultrasound Imparted Air-recoil Resonance method for acoustic power estimation.** Sreekumar Kaiplavil (Inst. of Cancer Res., 15 Cotswold Rd., Sutton SM2 5NG, United Kingdom, sreekumarkaiplavil@yahoo.co.in)

Recently, we have introduced a novel method, the Ultrasound Imparted Air-recoil Resonance (UIAR), for the estimation of acoustic power with high sensitivity. Salient features of this approach over existing practices include fast response, electrical and magnetic inertness and hence MRI compatibility, portability, high damage threshold, and immunity to vibration and interferences, low cost, etc. The angle of incidence should be fixed for accurate measurement. However, the transducer-detector pair can be aligned in any direction with respect to the force of gravity. In this sense, the operation of the device is orientation-independent. The device is useful in the case of pulsed/burst as well as continuous ultrasound exposure. Sensitivity was found to be extendable down to the micro Watt range or even below that, however, critical issues related to the thermo-viscous loss mechanisms in the system need careful optimization. The power handling capacity is a few hundreds of Watts, which could be augmented using suitable materials for the Helmholtz resonator window that functions as the acoustic filed sensing head. A detailed account of the sensitivity and high-power estimation capability of the UIAR technique will be presented in the paper.

**11:00**

**4aBA10. Algebraic reconstruction technique considering curved ray for sound-speed tomography with ring-array transducer.** Hirofumi Nakamura, Tetsuya Kanagawa (Dept. of Mech. Eng., The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, nakamura@fel.t.u-tokyo.ac.jp), Satoshi Tamano (Hitachi Aloka Medical, Mitaka, Japan), Takashi Azuma (Dept. of Mech. Eng., The Univ. of Tokyo, Bunkyo-ku, Japan), Kiyoshi Yoshinaka (Dept. of Human Life Technol., Adv. Industrial Sci. and Technol., Tsukuba, Japan), Akira Sasaki, Shu Takagi, and Yoichiro Matsumoto (Dept. of Mech. Eng., The Univ. of Tokyo, Bunkyo-ku, Japan)

Our objective is to develop an ultrasound treatment and diagnosis integrated system for breast cancer. Ultrasound computed tomography (UCT) in imaging and high intensity focused ultrasound (HIFU) in therapy was integrated to achieve ideal treatment system. Profiles of sound speed and attenuation obtained by UCT has informative parameters to correct deformation of HIFU beam. We try to develop an imaging system using ring-array transducer with 1024-elements, multiplexer connecting 1024 to 256 and Verasonics programmable imaging system with 256 channels. First, an iterative Simultaneous Algebraic Reconstruction Technique (SART) reconstruction methods with an assumption of straight path was employed. SART was applied to projection data calculated by a FEM simulator treating actual curved ray caused by tissue inhomogeneity. In these results, estimated error in sound speed difference was 53%. Then we introduced Linear Travel-time Interpolation (LTI) to SART to implement effects of curved ray. LTI is a ray tracing method based on Fermat's Principle. We evaluated the LTI and SART integrated method for sound-speed tomography. Travel-time for reconstruction was achieved from simulation based on finite difference method. The reconstruction image was highly corresponded with original image. Reconstructed image from experimental projection data will be reported in the presentation.

**11:15**

**4aBA11. Albumin based gel phantoms with controllable thermal sensitivity for quantifying ultrasound thermal energy.** Rei Asami and Kenichi Kawabata (Central Res. Lab., Hitachi, Ltd., 1-280 HIgashi Koigakubo, Kokubunji, Tokyo 185-0003, Japan, rei.asami.fq@hitachi.com)

Medical ultrasound safety is mainly concerned with two factors, thermal and mechanical. A simple method to indicate temperature rise by ultrasound is essential to standardize diagnostic and therapeutic techniques. A thermocouple is used for direct measurement but the effect of viscous heating is inevitable. MR thermometry provides accurate measurements but requires expensive machinery. Bovine serum albumin (BSA) gels are known to change colors from transparent to opaque at 70 °C and used as simple temperature indicator. We propose a new temperature indicator that combines additives with BSA gels to modify the denaturation temperature. Polyacrylamide gels were prepared with 9% BSA and various concentrations of additives such as ovarian albumin and gelatin. Gels were then placed in degassed water at 37 °C and adjacent to a focused ultrasound transducer ($f0 = 3.3$ MHz). Ultrasound irradiation process was video-recorded for image processing. Gels prepared with ovarian albumin resulted in ranges of denaturation temperature of 40—60 °C. Ovarian albumin, hardly soluble protein, could have lowered the stability of BSA. Recorded timing of the denaturation fitted well with the computer based simulation result. The result suggests that additives can widen the application of BSA gels in quantifying ultrasound thermal energy. [This study was, in part, funded by Japanese Menistory of Economy, Trade and Inductor.]

# Session 4aEA

## Engineering Acoustics: Energy Harvesting From Acoustic Phenomena

Kenneth Cunefare, Chair
*Georgia Tech., Mech. Eng., Atlanta, GA 30332-0405*

### *Invited Papers*

**8:30**

**4aEA1. Low-power electricity generation from dynamical systems.** Alper Erturk (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, alper.erturk@me.gatech.edu)

This talk will review our research on energy harvesting from electroelastic dynamical systems for low-power electricity generation with an emphasis on piezoelectric transduction. The transformation of vibrations into electricity using piezoelectric materials with the goal of powering small electronic components has received growing attention over the last decade. Enabling energy-autonomous small electronic components can lead to reduced maintenance costs in various wireless applications, such as structural health monitoring of civil and military systems. After a brief discussion of energy harvesting methods for low-power electricity generation, this talk will be focused on linear and nonlinear energy harvesting using piezoelectric materials through the topics of distributed-parameter electroelastic dynamics of energy harvesters, performance and frequency bandwidth enhancement by exploiting nonlinear dynamic phenomena, deterministic and stochastic excitation of monostable and bistable configurations, effects of dissipative and inherent electroelastic nonlinearities, electroaeroelastic flow energy harvesting using airfoil-based and bluff body-based configurations, and enhanced harvesting of structure-borne propagating waves using elastoacoustic mirrors and metamaterial structures. A brief introduction to our efforts on multi-functional underwater thrust and power generation using flexible piezoelectric composites will also be given.

**8:50**

**4aEA2. Energy harvesting from acoustic sources.** Kenneth Cunefare (Georgia Tech., Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

While energy harvesting from a variety of ambient sources (vibration, light, and wind) has been demonstrated and sensing and communication applications to exploit those sources have been developed, acoustic energy as an ambient source has not received much attention. The reason for this comes down to the basic physics of how much energy is available within an acoustic field. For airborne sounds, the energy density in sound fields that are perceived by humans to be quite loud (e.g., 80 to 160 dB, or ~0.2 Pa to ~2 kPa) actually represent an extremely low available energy source. In consequence, means must be taken to intensify an acoustic response, for example, through resonance, but even so, available energy remains limited. The exception to this issue in airborne sounds is the sound field that exists inside of an operating jet aircraft engine. The situation is quite different, however, when one considers pumped and pressurized fluid systems, where acoustic pressure variations due to the operation of pumps and other devices may reach into the mega-Pascal (MPa) range. Energy harvesting from such a fluid-borne acoustic source is feasible for powering sensors and wireless communication systems and has been successfully demonstrated.

**9:10**

**4aEA3. Aeroacoustic applications of acoustic energy harvesting.** Stephen B. Horowitz (Emerging Technologies Group, Ducommun Miltec, 678 Discovery Dr., Huntsville, AL 35806, shorowitz@one.ducommun.com) and Mark Sheplak (Dept. of Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

In this paper, the development and use of acoustic energy harvesting technology as a source of local power for aeroacoustic sensing and control applications is discussed. As an example, the application of acoustic energy harvesting as a primary local power source for aircraft engine noise reduction technology is addressed. Noise generated in turbofan engine nacelles can easily exceed 150 dBSPL, presenting a primary motivation for aircraft noise reduction technologies. Adaptive noise control approaches require less power than active methods and can outperform passive techniques (e.g., by actively tuning an otherwise passive system to a changing noise spectrum). Locally sourced power is highly desirable in this application to eliminate cabling in a difficult to access, harsh environment. The low power requirements can be reasonably supplied with harvested acoustic energy, particularly given the large acoustic intensities in and around aircraft engine nacelles. The detailed development approach and experimental results of acoustic energy harvesting using an electromechanical Helmholtz resonator will be presented. Additionally, alternative Helmholtz resonator variants and other aeroacoustic applications of acoustic energy harvesting will be reviewed.

## *Contributed Papers*

**9:30**

**4aEA4. Coupling efficiency analysis of hydraulic pressure energy harvesters.** Ellen Skow, Kenneth Cunefare, and Alper Erturk (Georgia Inst. of Technol., 620 Peachtree St. NE, Unit No. 1012, Atlanta, GA 30308, eskow3@gatech.edu)

The acoustic pressure within hydraulic systems, referred to as pressure ripple, is a high intensity energy source that can be utilized for powering sensor networks. A section of such a system can be modeled as a one dimensional waveguide, where the intensity can reach up to 1000 mW/cm$^2$ from a 300 kPa pressure ripple (peak-to-peak acoustic pressure) within a hydraulic system. Hydraulic pressure energy harvesters (HPEH) are devices designed to convert the pressure ripple into electrical energy, thereby enabling wireless sensor nodes. HPEH couple the dynamic fluid pressure to a piezoelectric stack, which is connected to a harvester circuit to optimize power output. A key aspect of the HPEH design is the fluid-mechanical coupling of the pressure ripple to the stack for maximizing the energy extracted. The efficiency of HPEH device and harvester circuit potential power output can be determined using the volume velocity of the pressure ripple, the coupling efficiency of the HPEH, and the conversion efficiency of the piezoelectric stack. In this work, the coupling efficiency and the power output efficiency of currently developed HPEH devices will be analyzed and compared to modeled efficiency of such devices.

**9:45**

**4aEA5. An effective theory for meta-mass and meta-material mechanical/electrical devices.** John J. McCoy (The Catholic Univ. of America, 1922 New Hampshire Ave., Washington, DC 20009, mccoy@cua.edu)

A dynamical system comprised of a rigid housing element that encapsulates a large multiplicity of oscillators is said to comprise a "meta-mass," in that there is no direct interaction of the external environment and the oscillators, the presence of which is inferred in an observed non-Newtonian housing element behavior. For a class of meta-masses, for which the oscillators have more-or-less equal masses with resonances that densely fill a frequency band more-or-less uniformly, the non-Newtonian behavior is most transparently seen in a different "energetics," i.e., the transfer of energy between the housing element and an external source, this occasioned by an inherently "transitory "internal energy transfer process. Encapsulating a large multiplicity of meta-masses in an ordinary material specimen is said to comprise a meta-material specimen, a somewhat different understanding in not having the direct interaction of each individual oscillator with every other individual oscillator obtain through the behavior of the material matrix. Constructed is a mathematical framework for quantifying limited observable system behavior while not accommodating all underlying physics, i.e., an "effective theory," this a prerequisite for the design of meta-mass and meta-material mechanical/electrical devices.

THURSDAY MORNING, 5 DECEMBER 2013 — SUTTER A/B, 9:30 A.M. TO 11:00 A.M.

# Session 4aID

## Interdisciplinary: Oral History Bootcamp

Victor Sparrow, Chair
*Grad. Program in Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802*

In this bootcamp, participants will learn appropriate techniques to conduct oral history interviews. No advanced reservations are required. Dr. Gregory Good is the Spencer Weart Director of the Center for the History of Physics of the American Institute of Physics (AIP), and he will coach the session participants in the nuts and bolts of preparing for, conducting, and following up after an oral history interview session. Dr. Good is very experienced with collecting oral histories.

If you are interested in the history of acoustics and in preserving that history, the ASA Committee on Archives and History invites you to participate in this bootcamp. Oral histories are a very important part of documenting the background and motivations for administrative and scientific contributions, the part of history that is not usually available in the printed record, such as peer-reviewed publications. So oral histories fill the gaps on why someone dedicated much of their professional life to a particular topic or describes the journey they traveled to reach notable goals and/or make lasting contributions to the field.

Your help is needed to preserve this history, the history of acoustics. Thanks for participating!

## Session 4aMU

## Musical Acoustics and Structural Acoustics and Vibration: Acoustics of Percussion Instruments I

Thomas Moore, Chair
*Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789*

### *Invited Papers*

**9:00**

**4aMU1. Percussion, performance, pedagogy, and technology: The impact of virtual music Lessons in percussion education.** Rohan Krishnamurthy (Musicology, Eastman School of Music, 544 Sunrise Circle, Kalamazoo, MI 49009, rohan.krishnamurthy@rochester.edu)

Real-time net-conferencing software such as Skype and Google Hangout has revolutionized music education in the last decade. This presentation examines the intersections of percussion, performance, pedagogy, and technology with specific reference to South Indian classical Carnatic percussion. Drawing on several scholarly perspectives and ethnographic research in India, various aspects of online teaching will be discussed, including its history and motivations, comparison of online and face-to-face pedagogical repertoire, negotiating audio/video lag or latency, and other significant social and cultural dimensions. Broader implications of online pedagogy in terms of reinforcing musical and intellectual centers, as well as strategies for improving the online learning experience, will be explored. A live, interactive performance on the mridangam, the principal percussion instrument from South India, will follow the presentation.

**9:20**

**4aMU2. The contribution of spectrum and tempo to auditory streaming of simple and compound "bols" in tabla rhythms.** Punita G. Singh (Sound Sense, 16 Gauri Apt., 3 Rajesh Pilot Ln., New Delhi 110011, India, punita@gmail.com)

Rhythms on the tabla, a north Indian percussion instrument, are generated by producing sounds on one or two drums separately or simultaneously to produce simple or compound "bols." At high speeds, auditory stream segregation based on spectral properties of adjacent bols can create parallel perceptual layers that can be leveraged strategically by percussionists. This observed phenomenon was studied experimentally by constructing sequences of bols in which adjacent sounds shared different spectral regions. For example, the bols "ghe" and "tin," which have very different spectra, were placed on either side of the compound bol "dhin," which contains both "ghe" and "tin." At a moderate tempo, the sequence is typically heard as a gallop rhythm. However, at quicker tempi, streaming occurs and components of the compound bol "dhin" group perceptually with their neighboring counterparts, to create parallel layers of pairs of "ghe," "ghe" and "tin," "tin", instead of the galloping triple. Listeners identified when streaming took place as a function of the specific bols and tempi used. Spectral analyses of the bols indicated that perceptual segregation was indeed based on proximity of spectral loci. At high speeds, spectral differences are perceptually highlighted and facilitate the formation of auditory streams.

**9:40**

**4aMU3. Measurements of coupled drumhead vibrations using electronic speckle-pattern interferometry.** Randy Worland and Benjamin Boe (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Many musical drums such as snare drums, tom toms, and bass drums consist of two membranes at opposite ends of a cylindrical shell. Striking one head causes both to vibrate, as their motions are coupled due to the enclosed air as well as the shell itself. An optical system consisting of two electronic speckle-pattern interferometers was constructed allowing operating deflection shapes of both heads to be viewed simultaneously while the drum is driven acoustically at a resonant frequency. This system allows the determination of the relative phases, orientations, and amplitudes of the vibrational patterns on the two heads. Previously reported results for coupled drumheads were verified and extended to include the effects of degenerate single membrane mode pairs that are split due to non-uniform tension in the heads. Examples of higher frequency coupled patterns are also shown. Parameters influencing the degree of coupling are described briefly and compared qualitatively with results from a finite element model.

**10:00**

**4aMU4. Expressively actuated percussion instruments and interfaces.** Edgar J. Berdahl (Music, Louisiana State Univ., 102 New Music Bldg., Baton Rouge, LA 70803, eberdahl@ccrma.stanford.edu)

Percussion instruments and interfaces can be extended by incorporating actuation with motors. While many prior designs have employed actuation to simply trigger notes, this work emphasizes the importance of the musician's gesture using expressively controlled actuators. This is achieved by implementing force-feedback control of the actuators. For example, the Haptic Drum essentially consists of a drum pad attached to a woofer. Every time that a drumstick strikes the drum pad, the woofer briefly pushes it upward again, adding energy to the motion of the drumstick. In this manner, the Haptic Drum enables musicians to play one-handed drum rolls at optionally superhuman speeds and with arbitrarily complex dynamics. Furthermore, because the musician and the drumstick are inside the feedback control loop, the musician can change the rate of the drum roll by modulating the stiffness of his or her muscles or changing the downward force applied to the drumstick. Conversely, borrowing on technology for remote surgery, expressively actuated percussion

instruments can be designed using force-feedback teleoperation. Simply by recording the position of the master haptic device during teleoperation, generalized motor programs can be created for expressively playing percussion instruments. This technology was employed in the composition “When the Robots Get Loose."

### *Contributed Papers*

**10:20**

**4aMU5. Teponaztli, an ancient percussion instrument.** Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

Ancient percussion instruments in Mesoamerica included the Huehuetl and the Teponaztli, both made out of a single piece of hollow trees, which pretty often were decorated with low profile representations of animals, warriors or other ceremonial symbols, both were often used by the Aztecs and Tarascas among many other cultures in the region. The first one is normally used in a vertical position and has an strained animal skin, typically an ocelot, in the top side in order to excite it with wooden sticks or by the bare hands, while the second is formed by a couple of tongues made out from the same tree cortex, and excited by a couple of wooden sticks covered with hard rubber. In both the hollowed tree forms a resonant chamber in order to increase the sound level, and include openings in order to allow the exit of the sound from inside of the drum. They were employed for ceremonies, festivities and communication. Here, there are presented the sound characteristics of a small “teponaztli,” which is actually a small copy scaled about 5:1 of a real one, which can usually only be seen in archaeological museums.

**10:35**

**4aMU6. Acoustical measurements and finite difference simulation of the West-African “talking drum".** Florian Pfeifle (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstrasse 13, Hamburg 20354, Germany, Florian.Pfeifle@uni-hamburg.de)

A very iconic drum, commonly found in West-Africa, is the so called “talking drum.” It has a large geographic distribution, spreading several countries and ethnic groups. The characteristic feature of this unique instruments is the membrane, which can be tuned, through varying the tension of the membrane fixture, while playing. This effect can be utilized to play melodies and pitch-glides on the drum. West-African musicians use this effect to mimic speech patterns and speech melodies from tonal languages. This highly non-linear excitation of the drum leads to several questions regarding the acoustical properties of the instrument. The acoustical research of the drum include high-speed measurements of the drum-head to quantify the changing tension distribution over the membrane. Microphone array measurements, with 128 microphones, are applied to examine the radiation characteristics of the instrument. A special focus is put on the resonance frequencies of the enclosed air volume and the influence it has on the top and bottom membrane as well as the damping characteristics of the drum head. A physical model of the instrument, including a Kirchhoff-Carrier-like tension modulated, 2-dimensional, orthotropic, stiff membrane, coupled to a 3-dimensional air-volume is implemented and compared to the measurements.

**10:50**

**4aMU7. Acoustic properties of carbon fiber in percussive instruments.** Alex Wion, Rustin Vogt, and Patrick Homen (Mech. Eng., California State Univ. Sacramento, 1255 University Ave., Apt. 117, Sacramento, CA 95825, wion.07@gmail.com)

With advancements in manufacturability and increasing accessibility of materials, composite materials are competing with traditional materials in the design and manufacturing of musical instruments. Specifically, in snare drum applications, carbon fiber encased in an epoxy matrix has been chosen for its physical properties of strength, durability, and weight as well as its psychoacoustic properties compared to wood. Through comparative data acquisition analysis, the acoustic properties of composite snare drum shells are examined against traditional wooden snare shells. By subjecting the shells to controlled audio frequencies, the sound energy response on the air medium outside of the shell was measured and compared to wooden shells. Experimental data were examined was compared to theoretical values found for carbon fiber.

THURSDAY MORNING, 5 DECEMBER 2013 — CONTINENTAL 9, 8:00 A.M. TO 9:45 A.M.

## Session 4aNS

## Noise: Outdoor Sound Propagation and Modeling

Lauren M. Ronsse, Chair
*Audio Arts and Acoust., Columbia College Chicago, 33 E. Congress Pkwy, Ste. 601, Chicago, IL 60605*

### *Contributed Papers*

**8:00**

**4aNS1. Comparison and evaluation of physics-based outdoor sound propagation assessment schemes.** Lauren M. Ronsse (Audio Arts and Acoust., Columbia College Chicago, 33 E. Congress Pkwy, Ste. 601, Chicago, IL 60605, lronsse@colum.edu) and Dan Valente (Construction Eng. Res. Lab., Engineer Res. and Development Ctr., Champaign, IL)

Evaluating the effectiveness of outdoor sound propagation assessment schemes using experimentally collected data is a complex task. Specifically, when the acoustic data collected assume non-Gaussian distributions with differing sample sizes per assigned class, traditional parametric statistical techniques may not be used. As an alternative, this research introduces an original cost function to evaluate and compare various methods aimed at defining acoustic propagation classes based on meteorological measurements. Class assignments in each scheme are based on the atmospheric stability, the strength of the vertical effective sound speed gradient, and the vertical effective sound speed profile. The acoustic data included in this analysis were generated by a high-energy impulsive source and gathered at source-to-receiver distances of 1, 2, 4, and 8 km. The results indicate that the assessment scheme based on the strength of the effective sound speed gradient most effectively classifies the peak level sound propagation sampled in temperate and desert climate conditions at these distances.

**8:15**

**4aNS2. Using simplified terrain and weather mapping in outdoor sound propagation predictions.** Whitney Coyle, Victor Sparrow (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wlc5061@psu.edu), and Bruce Ikelheimer (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

A complementary experimental and computational study was undertaken to assess the variability due to model sensitivities when predicting propagation in realistic outdoor environments by using a Green's Function Parabolic Equation (GFPE) method and including realistic weather and terrain profiles. In order to test the validity of the basic and enhanced model including real terrain and weather data, field measurements were conducted at Hogan's Mountain, North Carolina. By incorporating USGS terrain and measured weather profiles, this project further developed a Hogan's Mountain specific GFPE to include a stair-step terrain mapping capability and linearly interpolated sound speed profiles. This Hogan's Mountain GFPE uses matching conditions for comparison to the measured received levels from the field test data. Due to the vast data set acquired, which allows for comparison of the model to measurement in a multitude of propagation situations, several comments are made regarding the choice of calculation parameters as well as the validity of the basic and altered Hogan's Mountain GFPE for different propagation prediction settings. Sample results will illustrate the agreement based on frequency, specific terrain and weather measured along the propagation path for individual source-receiver pairs of interest. [Work supported by Spawar Systems Center Pacific.]

**8:30**

**4aNS3. Data-driven prediction of peak sound levels at long range using sparse, ground-level meteorological measurements and a random forest.** Dan Valente (US Army Engineer Res. & Development Ctr., PO Box 9005, Champaign, IL 61826, daniel.p.valente@usace.army.mil)

Outdoor sound propagation is highly dependent upon meteorological conditions. While this, of course, is a trivial statement, predicting sound levels based on meteorology is not. This is especially true for signals that propagate many kilometers, as is the case for those generated by high-energy impulsive sources such as explosions and heavy weaponry; waves have ample opportunity for refraction by and scattering from local atmospheric features along the entire propagation path. The range of received blast levels at distances greater than 2 km can span nearly 50 dB, depending on weather conditions. Using a statistical learning method known as a Random Forest, we demonstrate the prediction of levels from simple meteorological measurements in the face of this extreme variability. With simple, spatially sparse meteorological data, the model can predict levels to within 3 dB at 2 km and 5 dB at 15 km. The results presented here suggest that as more data are acquired through continuous noise monitoring programs, physics-blind, data-driven statistical models have the potential to supplant computationally intensive propagation models for noise prediction. Caveats and cautions when using these types of machine learning methods will also be discussed.

**8:45**

**4aNS4. Variability in acoustic transmission loss over a rough water surface.** Cristina Tollefsen and Sean Pecknold (Defence Res. and Development Canada - Atlantic, P. O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, cristina.tollefsen@gmail.com)

Variability in the acoustic transmission loss of impulsive sounds propagating in the atmosphere over a rough water surface was measured over time scales of 1–2 h at a fixed range of 250 m. On two separate days (26 and 30 May 2012), a propane cannon source was deployed on the upper deck of a ship. Once per minute, the propane cannon fired volleys of four shots that were recorded on a receiver deployed on a small boat tethered upwind of the ship. Meteorological conditions and sea state were comparable on both days, resulting in similar observations for transmission loss: mean and standard deviation of 64 $\pm$ 5 dB (26 May) and 66 dB $\pm$ 4 dB (30 May). The variability in transmission loss was high, with minimum and maximum values of 48 and 75 dB (26 May) and 53 and 77 dB (30 May). The transmission loss measured throughout the experiment exceeded the 48 dB predicted by assuming spherical spreading, since the receiver was upwind of the source. Measured results are compared to transmission loss computed from a parabolic equation model using an ensemble of turbulence and rough sea surfaces estimated from the meteorological conditions on board the ship.

**9:00**

**4aNS5. Mapping the extent of noise on a national scale using geospatial models.** Dan Mennitt, Kurt Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, daniel_mennitt@partner.nps.gov), Lisa Nelson (Inventory and Monitoring Div., National Park Service, Fort Collins, CO), and Megan McKenna (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

Because many wildlife habitats, geological processes, and anthropogenic impacts occur on a regional scale, acoustical analyses must encompass a similar extent. Geospatial sound modeling incorporates spatial representations of biological, geophysical, climatic, and anthropogenic factors to assess expected contributions to the existing sound pressure level from both anthropogenic and natural sources. This method enables mapping of sound pressure levels at national scales. The models do not directly apply the physics of sound propagation or characteristics of individual sound sources. Instead, long-term sound pressure level measurements from hundreds of sites across the contiguous United States were used to train regression models to predict acoustic conditions. This talk will focus on the implications of the resulting acoustic maps of the contiguous United States. In addition, noise and the relationship to light pollution will be considered.

**9:15**

**4aNS6. Intensity analysis of peak-frequency region in noise produced by a military jet aircraft.** Trevor A. Stout, Kent L. Gee, Tracianne B. Neilsen, Alan T. Wall (Phys., Brigham Young Univ., 688 North 500 East, Provo, UT 84606, titorep@gmail.com), and Michael M. James (Blue Ridge Res. and Consulting LLC, Asheville, NC)

Acoustic intensity measurements of the F-22A Raptor are analyzed as part of ongoing efforts to characterize the noise radiation from military jet aircraft. Data were recorded from a rig of microphones and an attached tetrahedral intensity probe at various locations to the sideline and aft of the aircraft. Recently, techniques such as coherence, similarity spectra analyses, and near-field acoustical holography have indicated a peak-frequency region comprised of two maxima that have very different radiation directionalities. Acoustic vector intensity is analyzed as a function of frequency to further assess the behavior of this double-peak phenomenon, which is not accounted for by current jet noise models. The results thus far confirm the discrete nature of the peaks and their directionalities.

**9:30**

**4aNS7. Source motion modeling for high-speed aircraft noise propagation.** Bao N. Tong and Kai Ming Li (School of Mech. Eng., Purdue Univ., 140 S. Martin Jischke Dr., West Lafayette, IN 47907-2031, bntong@purdue.edu)

A continuous source motion model has been developed to represent a cruising aircraft traveling at high altitudes. The numerical implementation is based on the Lorentz transform (LT) and a Fast Field Program (FFP) formulation, which is used to compute the sound fields due to a monopole point source traversing in a horizontally stratified atmosphere parallel to a ground surface. To reduce the computational expenses, a one-dimensional LT-FFP is performed to obtain the pressure time-history for overhead flight conditions. The continuous source motion model is compared against a heuristic model which applies a Doppler shift to the stationary source sound field. A linear sound speed profile was selected to simplify the ray model implementation. A parametric study involving the source Mach number and source emission frequency has been performed in a variety of environmental conditions. The differences in the predicted maximum sound pressure levels between the two models can be as large as 9 dB under certain conditions. Numerical simulations indicate that low source emission frequencies (e.g., 50–300 Hz) combined with high source Mach numbers tend to result in larger discrepancies. The numerical simulations suggest the importance of including the effects of convective source amplification, especially, for turboprop aircraft noise propagation.

# Session 4aPA

## Physical Acoustics: Advances in Infrasound Research I

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

Daniel Kleinert, Cochair
*National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677*

### *Invited Papers*

**8:00**

**4aPA1. Infrasonic wind noise in a pine forest.** Richard Raspet, Jeremy Webster, and JohnPaul Abbott (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, raspet@olemiss.edu)

Comparison of the ambient infrasonic noise levels at 39 stations of the International Monitoring System indicated that wind noise levels are as much as 6 dB quieter at heavily vegetated stations. This combined experimental and theoretical study investigates mechanisms for wind noise generation in and above a uniform pine forest (h ~7.0 m). A 10 m tower instrumented with ultrasonic anemometers measures the turbulence field and wind velocity profile above and below the canopy and an infrasonic transducer is collocated on the ground. A prediction of the turbulence-shear wind noise contribution from the turbulence above the canopy and also from the turbulence within the canopy are calculated and compared to the data. The above canopy contribution agrees well with the data for the low frequency regime (<0.5Hz). Surprisingly, this contribution is larger than would occur over a grassy plane with similar meteorological conditions. The under canopy turbulence-shear interaction calculation underpredicts the small high frequency contribution, however an estimate of the turbulence-turbulence interaction wind noise in the under canopy layer provides a good fit to this data. [Work supported by the U. S. Army Research Laboratory and the U. S. Army Research Office under grant W911NF-12-1-0547.]

**8:20**

**4aPA2. New results exploiting correlation in wind noise to enhance detection of transient infrasound.** William G. Frazier (Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, frazier@olemiss.edu)

A previous ASA presentation introduced a signal processing method for exploiting the correlation observed in pressure fluctuations measured by infrasound sensors on small spatial-temporal scales in order to achieve enhanced signal detection and estimation performance. An algorithm based on representing the pressure field with Fourier series and finite-element bases was presented and applied to time series data in the 0.1 to 1.0 Hz band with inter-sensor spacing of 1 m using small domes as wind screens. The coherence of the data was not inspiring, but the performance was consistent with this low coherence. In this presentation new data collected using flush-to-the-earth installed sensors spaced 1 and 0.5 m apart will be presented and analyzed. In addition, application of a generalized least-squares approach to estimating the infrasound signal will be examined and compared to the previous method. Results demonstrate significant improvements over previous results at frequencies below 1.0 Hz.

**8:40**

**4aPA3. Calibration and characterization of the response of infrasound sensors to environmental factors.** Carrick L. Talmadge (NCPA, Univ. of Mississippi, 1 Coliseum Dr., Unviersity, MS 38677, clt@olemiss.edu)

An infrasound calibration system has been developed at the National Center for Physical Acoustics. The calibration tank is comprised of a 1 in. cylindrical shell 40-in. in diameter, 40-in. long, with 40 in. hemispherical end caps. The interior volume of the tank is approximately 1.8 cubic meters. Up to eight normal-sized infrasound sensors can be enclosed in the volume for one measurement session. Pressure and temperature in the interior and exterior of the tank are also monitored. The pressure source is a 10-in. driver, which is calibrated using multiple barometers that are placed internal to the tank. The measurement paradigm is to drive the tank at a constant amplitude (typically 10-Pa peak-to-peak) and known frequency (typically 0.5 Hz), and to track the variation in the measured response of the test sensor with respect to multiple reference sensors in order to characterize the effect of manipulation of the environment (static pressure, temperature, and seismic motion) on the test sensor. For sensors with very low nonlinearity, allowing the static pressure to change while the driving amplitude of the speaker was held constant was found to be necessary for assessing the effect of nonlinearity on these sensors. Variations in sensitivity on the order of 50 ppm can be measured in a 10-s interval.

9:00

**4aPA4. Passive acoustic remote sensing and anomalous infrasound propagation studies.** Jelle D. Assink (DAM/DIF/DASE, CEA, Bruyeres-le-Chatel, Arpajon F-91297, France, jelle.assink@gmail.com), Roger Waxler (NCPA, Univ. of Mississippi, University, MS), Laslo Evers, Pieter Smets (KNMI, De Bilt, Netherlands), Alexis Le Pichon, and Elisabeth Blanc (DAM/DIF/DASE, CEA, Bruyeres-le-Chatel, France)

In this talk, we will present recent work on various infrasound remote sensing studies. We will focus on bi-directional stratospheric ducting during a Sudden Stratospheric Warming (SSW) event and the associated infrasonic signature. We present infrasound data in which the described effect is captured with microbarom signals in the Mediterranean region. Microbarom source locations are modeled using operational ocean wave models. The modeling reveals a previously unidentified microbarom source region in the Eastern Mediterranean besides the more typical microbarom source region in the Atlantic Ocean. This work illustrates that the classic paradigm of a uni-directional stratospheric duct for infrasound propagation can be broken during a SSW event. Furthermore, we will present a case study in which the influence of atmospheric dynamics on infrasound propagation is studied. We make use of over 6 years of nearly continuous volcanic infrasound recordings from Mount Etna, Italy (37 N) that are available through the Atmospheric dynamics Research InfraStructure in Europe (ARISE) network. The infrasound observables are compared to theoretical estimates obtained from propagation modeling using existing European Centre for Medium-Range Weather Forecasts (ECMWF) atmospheric databases. While a good agreement is often found, we also report on significant discrepancies around the equinox period and during intervals during which anomalous detections occur during the winter.

## *Contributed Papers*

9:20

**4aPA5. Refraction of impulsive signals by a mountain slope.** Roger M. Waxler (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu) and Doru Velea (SAIC, Reston, VA)

As part of a recent experiment on signals from blast waves, it was possible to instrument a mountain slope to the east of the source locations. Four infrasound sensors were deployed about 1 km apart from the base of the mountain up. In addition, infrasound sensors were deployed close to the mountain peak, and then along a ridge extending to the east at roughly constant altitude. It was observed that signals developed a long, low frequency tail as they propagated up the slope. Data and theoretical analyses will be presented.

9:35

**4aPA6. Ground and aerostat measurements of wind noise in the infrasonic range.** W. C. Kirkpatrick Alberts (US Army Research Lab., 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net), Roger Waxler (Univ. of Mississippi, University, MS), Christian Reiff, and Leng Sim (US Army Research Lab., Adelphi, MD)

Conventional deployment of infrasound sensors typically requires the sensors to be on the ground with extensive wind screens, e.g., porous hose or pipe arrays, in order to minimize wind induced noise on the sensor. Thus, flying an infrasound sensor on a balloon would seem ill-advised because of the inability to carry sufficient wind screens to mitigate the increased winds aloft. However, during an experiment designed to monitor short-range characteristics of low-frequency impulsive events, an infrasound sensor was placed aboard a tethered aerostat while the balloon flew to a maximum height of approximately 300 m. Comparisons between noise levels at the airborne sensor and at a nearby ground based sensor will be discussed. Noise levels aboard the aerostat are found to be similar to those on the ground.

9:50

**4aPA7. Direct measurement of acoustic impedance for wind-noise-reduction pipe systems and components.** Chad M. Smith and Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., PO Box 30, State College, PA, cms561@psu.edu)

A wind-noise-reduction pipe system can have a significant effect on the frequency response of an infrasound array element especially above a few tenths of a hertz. If there is a defect in the pipe system—a clogged resonance suppressor, a blocked or broken pipe, or a flooded manifold, for example—measurement of the frequency response may indicate the presence of the defect but not necessarily the type or location. Direct measurement of the acoustical impedance made at accessible points in the pipe system may be a useful adjunct to *in-situ* response measurement for identification of the type and location of defects, which may expedite repair especially if most of the system is buried. A prototype of a portable impedance instrument has been constructed and tested both in the lab and under field conditions. Measurements of individual pipe-system components—inlets, pipes, manifolds, resonance suppressors, or gravel piles—are shown along with comparison to theoretical predictions. In addition, defects are intentionally introduced in a rosette pipe system to determine the sensitivity of the acoustical impedance measurement to such defects.

10:05–10:20 Break

10:20

**4aPA8. Experimental investigation of large porous wind fences for infrasonic wind noise reduction.** JohnPaul R. Abbott, Richard Raspet, and Jeremy Webster (Dept. of Phys. and Astronomy, National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., Rm. 1044, Oxford, MS 38677, johnpaul.abbott@gmail.com)

An extensive experimental investigation of large porous wind fences constructed from commercially available materials has recently been completed. Measured changes to the wind noise, turbulence, and wind velocity profile inside the fence resulting from varying the fence's height, diameter, and porosity have been measured. The effect of other variables, including porous roofs, a bottom gap, and the addition of secondary windscreens were also studied. An empirical model based on measurements of the turbulence correlation length around the outside of the fence and the velocity gradient across the fence's surface has been derived to develop a better understanding of the wind noise reduction mechanisms. The measurements and model are then used to suggest an optimized wind fence design and predict the band width and magnitude of the wind noise reduction curve.

10:35

**4aPA9. On the interaction of infrasonic waves with internal gravity waves perturbations.** Jean-Marie Lalande (Jamie Whitten National Ctr. for Physical Acoust., NCPA, 1 Coliseum Dr., University, MS-38677, jean-marie.lalande@gmail.com), Roger Waxler, and Joel Lonzaga (Jamie Whitten National Ctr. for Physical Acoust., NCPA, Oxford, MS)

Infrasonic waves propagate at long range through atmospheric ducts resulting from the stratification of atmospheric properties. These ducts are characterized by their spatio-temporal variability. Hence, infrasonic waves integrate atmospheric information along their propagation paths. In order to study infrasonic wave propagation, we resort to atmospheric specification combining Numerical Weather Prediction and climatological models. However, due to the lack of observations, these models fail to describe small scale variability such as perturbations associated to the presence of internal gravity waves. These waves play an important role in the atmospheric dynamic by transferring momentum to the mean flow at critical levels and at wave-breaking altitudes. In this study we intend to describe the interaction of infrasonic waves with internal gravity waves in order to understand

the long-tail behavior observed in infrasound broadband signals. We developed a model for the propagation of internal waves used to generate realistic perturbations of the background atmospheric states. By using a linear full-wave model of infrasound propagation, our goal is to ultimately relate infrasound characteristics to internal waves properties.

**10:50**

**4aPA10. Preliminary study of infrasonic attenuation and dispersion in the lower thermosphere, based on non-continuum fluid mechanics.**
Akinjide Akintunde and Andi Petculescu (Univ. of Louisiana at Lafayette, PO Box 44210, Lafayette, LA 70504, akin@louisiana.edu)

A framework for predicting thermospheric attenuation and dispersion of infrasound between 80 and 160 km will be presented. The work is part of a pilot study whose goal is to complement the currently established models of thermospheric propagation, the standard being the Sutherland-Bass (SB) model [J. Acoust. Soc. Am. **115**(3), 1012–1032 (2004)], which overestimates the observed attenuation noticeably. Based on the Navier-Stokes equation and its associated momentum and heat fluxes, the SB model treats the higher atmosphere as a continuum. However, for a given wavelength, the Knudsen (Kn) number increases rapidly in the thermosphere due to the high mean free path gradient. For Kn > 0.01, the continuum Navier-Stokes equations are no longer accurate. The present work shows how the predicted wavenumber changes when non-continuum approximations (e.g., the Burnett and 13-moment equations) are used. The ambient parameters and thermophysical properties are extracted from NIST, at the partial pressures of the main constituents ($N_2$ and $O_2$). The effects of rotational relaxation, gravity and tidal waves, neutral-charged and charged-charged particle interactions, UV heating/cooling rates, and other thermospheric processes will be addressed. [The work was funded by NSF-EPSCoR/Louisiana Board of Regents.]

**11:05**

**4aPA11. An exact solution of a Burgers' equation governing the nonlinear propagation of infrasound in a range-dependent, windy atmosphere.**
Joel B. Lonzaga and Roger Waxler (National Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, jblonzag@olemiss.edu)

We present the derivation of an exact solution of an inviscid Burgers' equation that governs the nonlinear propagation of infrasound in a range-dependent, windy atmosphere with arbitrary sound speed, wind speed, and density profiles. The Burgers' equation is a reduced form of a nonlinear transport equation obtained using weakly, nonlinear ray theory. The solution extends known solutions for a homogeneous, windless medium to include the effects of wind and the inhomogeneity in the properties of the atmosphere. Analytical expressions for the shock velocity and period lengthening are readily obtained from the solution. The exact solution is used to validate our numerical algorithm which uses a spectral method to integrate the Burgers' equation. These models are compared with observed infrasound arrivals that clearly demonstrate nonlinear pulse steepening and stretching while propagating in the upper atmosphere.

# Session 4aPP

## Psychological and Physiological Acoustics: The Ear Club: Honoring Ervin R. Hafter and His Contributions to the Study of Binaural Processing and Auditory Cognition I

Neil F. Viemeister, Chair
*Dept. of Psych., Univ. of Minnesota, Minneapolis, MN 55455*

**Chair's Introduction—8:55**

### *Invited Papers*

**9:05**

**4aPP1. Selective attention and auditory filters.** Robert S. Schlauch (Speech Lang. Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, 115 Shevlin Hall, Minneapolis, MN 55455, schla001@umn.edu)

Erv Hafter has made significant contributions to our understanding of role of selective attention in detection tasks. I was fortunate to learn from and collaborate with Professor Hafter and his students on some projects addressing this topic as a post-doctoral fellow in his laboratory. We used the probe-signal method to measure auditory filters to learn about the allocation of listening bands under various cuing conditions. Our experiments demonstrated that auditory filters are under a listener's cognitive control. Our results also helped to explain why large losses are not observed in simple, tonal frequency uncertainty experiments. In Schlauch and Hafter (1991), an analysis of filter widths as the number of cue tones (monitored bands) increased supported an earlier finding by Hafter and Kaplan (1976) that auditory filters widen as the cost of shared attention increases. More recently, in a study inspired by my time in Hafter's lab, I completed an informational masking experiment that revealed individual differences in the liability of auditory filters that explains the poor performance of some persons in an informational masking task.

**9:25**

**4aPP2. Auditory informational masking and the Ear Club connection.** Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin, 1410 E. Skyline Dr., Madison, WI 53705, ralutfi@wisc.edu)

Erv Hafter has always made a special effort to promote recognition of young researchers in our field. His Ear Club of 40-plus years has provided a conversant audience for new investigators to test new ideas and his attendant hospitality has helped forge relationships valued throughout one's career. In this talk, I will describe how my own work benefited from visits with Erv. The focus is on informational masking (IM). I will present recent work suggesting a strong connection between two major, but seemingly unrelated, factors associated with IM: masker uncertainty and target-masker similarity. Experiments involving multitone pattern discrimination, multi-talker word recognition, sound-source identification, and sound localization are described. In each case, standard manipulations of masker uncertainty and target-masker similarity (including the covariation of target-masker frequencies) are found to have the same effect on performance provided they produce the same change in the information divergence of target and masker, a measure of statistical separation between signals from information theory. The results seem to reflect a general perceptual principle that segregates signals based on differences in their statistical structure. Future plans are to test the generality of this result in a simulated-cocktail-party environment [Hafter *et al*. Intl. Symp. Hearing, in press].



**9:45**

**4aPP3. Encoding of amplitude modulation upon interference in remote frequency regions.** Yi Shen and Virginia M. Richards (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697, shen.yi@uci.edu)

The modulation detection threshold for a sinusoidally amplitude modulated target tone was measured in the presence of another amplitude modulated interference tone. The interferer had a carrier frequency of 700 Hz and a modulation rate of 40 Hz. The target carrier frequency was 1300 Hz, and its modulation rate was 23, 33, 43, 63, and 80 Hz in separate conditions. During the psychophysical experiment, the auditory frequency following response (FFR) phase-locked to the 700-Hz carrier and the auditory steady-state response (ASSR) phase locked to the 40-Hz modulator were recorded. It was hypothesized that when greater interference occurs (when the interferer and target had closer modulation rates), the neural representation to the interferer might be enhanced or suppressed to a lesser degree. Neither the FFR nor ASSR demonstrated dependency on the target modulation rate, and no significant correlation was found between the behavioral and these two types of physiological measures. Results suggest the possibility that the modulation detection interference might originate from a location that is more central to the neural structures that give rise to FFR and the 40-Hz ASSR.

**10:05**

**4aPP4. Division of processing resources in auditory judgments.** Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov), Anne-Marie Bonnel, and Ervin R. Hafter (Dept. of Psych., Univ. of California, Berkeley, Berkeley, CA)

The attention operating characteristic (AOC) displays joint performance in a dual-task paradigm. Sampling theory allows the AOC to be used to distinguish two tasks that share resources from two tasks that call upon independent resources. Work in the Hafter lab over the past twenty years has examined the division of resources from several different perspectives. This talk will review data on intensity discrimination and identification showing that "easier" tasks can cause dual-task costs, while "harder" tasks can have no costs associated with the dual-task. Recent data on intensity discrimination and identification will be examined in the context of the division of processing resources, showing that the costs of sharing resources can increase with age of the listener. Finally, suggestions will be made for ways in which the current enthusiasm for the dual-task paradigm in clinical research can be used to improve our theoretical understanding of attention and memory as well as demonstrating the effects of various impairments and prostheses.

**10:25–10:40 Break**

**10:40**

**4aPP5. Revisiting Hafter and De Maio: How precision of coding of interaural delay varies with both magnitude of interaural delay and center frequency.** Leslie R. Bernstein and Constantine Trahiotis (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, 263 Farmington Ave., Farmington, CT 06030, Les@neuron.uchc.edu)

Erv Hafter's contributions span a broad range of topics fundamental to the understanding of monaural and binaural auditory perception. His publications and presentations are typified by historically important empirical, conceptual, and theoretical analyses. One issue he studied concerned how precision of neural coding of interaural time delay (ITD) varies with the magnitude of the delay and the spectral content of the stimulus. Such information remains central to contemporary quantitative models of binaural processing. Extending Erv's efforts, we have developed a new way to measure precision of ITD-coding as a joint function of ITD-magnitude and center frequency. The novel twist entails transforming the classic NoS$\pi$ condition into (NoS$\pi$)$\tau$ by imposing an ITD on the entire signal-plus-masker waveform. With that stimulus, uniform internal compensation of external ITDs would yield thresholds both independent of ITD and equal to those obtained under NoS$\pi$. In accord with the results of Hafter and De Maio [J. Acoust. Soc. Am. **57**, 181–187 (1975)], however, thresholds increased with ITD and did so more rapidly at 4 kHz than at 500 Hz. Our data were accounted for by assuming an internal, interaurally uncorrelated "processing noise," the power of which increases exponentially with the magnitude of the internal, "matching," delay.

**11:00**

**4aPP6. Lateralization, discrimination, detection, and Ervin.** Richard M. Stern (Dept. of Elec. and Comput. Eng., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, rms@cs.cmu.edu)

Erv Hafter's early work focused on the development of a comprehensive understanding of the binaural system, combining a characterization of the subjective phenomena associated with binaural lateralization with the results of objective experiments measuring interaural discrimination and binaural detection. To a large extent, this author's attempts to characterize a broad range of auditory phenomena based on a Jeffress-Colburn cross-correlation-based model can be viewed as a reformulation of the stimulus-based lateralization model that Hafter developed to provide a unified theoretical framework for his own findings. Erv Hafter has a gift both for designing clever and revealing experimental stimuli and paradigms, as well as for developing provocative interpretations of his experimental results. This paper will discuss a selection of complex binaural phenomena revealed by Erv's research in the context of their impact on contemporary theories of binaural interaction, as well as their impact on computational models of speech processing for robust speech recognition. [Work supported by DARPA and Cisco Research.]

**11:20**

**4aPP7. On the temporal weighting of binaural cues: precedence effects, rate limitations, and binaural adaptation.** G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232-8242, g.christopher.stecker@vanderbilt.edu), Andrew D. Brown (Physiol. and Biophys., Univ. of Colorado School of Medicine, Aurora, CO), Anna C. Diedeisch (Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN), and Jacqueline M. Bibee (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Over the course of three decades, Erv Hafter and his colleagues have investigated the relative effectiveness of binaural cues carried by different temporal portions of brief, mainly, high-frequency, stimuli. Those studies pioneered the use of filtered impulse trains (avoiding many limitations of sinusoidal amplitude modulation, and presaging the use of "transposed tones" for delivering envelope ITD), and revealed rate-limited processing of post-onset cues (Hafter and Dye 1983). Careful modeling of the effects on both ITD and ILD later led Hafter *et al.* (1988) to propose a pre-binaural origin for this binaural adaptation, and Hafter and Buell (1990) to present a simple quantitative model of monotonic, onset-triggered reduction of cue effectiveness over time. Erv's later students adopted observer-weighting paradigms to more directly measure the influence of each click in a train (Saberi 1996, Stecker and Hafter 2001), revealing some important and cue-dependent non-monotonicites. Our own work has continued this line of research, most recently investigating the temporal weighting of binaural cues in lower-frequency sounds. Consistent with a large body of Hafter's work, the results reveal an important temporal asymmetry in the effectiveness of binaural cues at both low and high frequencies and a key role for pre-binaural mechanisms. [Work supported by R03DC009482, R01DC011548.]

11:40

**4aPP8. Temporal effects when localizing targets defined by spatial consistency: Relation to Hafter's work on "binaural adaptation".** Robert H. Gilkey (Dept. of Psych., Wright State Univ., Dayton, OH 45435, gilk@wright.edu), Brian D. Simpson (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH), Eric R. Thompson (Ball Aerosp. & Technologies Corp., Fairborn, OH), Nandini Iyer, and Grffin Romigh (Battlespace Acoust., Air Force Res. Lab., Wright-Patterson AFB, OH)

Simpson *et al.* [Proceedings of Meetings on Acoustics **19**, 050140 (2013)] measured localization accuracy for sequences of noise bursts masked by simultaneous sequences of noise burst maskers (1 or 2). Targets were distinguishable from the maskers only in that all bursts within a target sequence arose from the same location, whereas each burst in a masker sequence arose from a different, randomly selected location. In this task, the spatial information extracted from each burst both helps define which element is the target ("target identification") and where that target is located ("target localization"). Presumably, the listener is more focused on target identification early in the sequence and on target localization later in the sequence. And so, it is not surprising that localization accuracy increases dramatically as the length of the sequences is increased. This talk will compare these results to the foundational work of Hafter and his colleagues [e.g., Hafter, 1997, in *Binaural and Spatial Hearing in Real and Virtual Environments*, edited by R. H. Gilkey and T. R. Anderson (Erlbaum, Mahwah, NJ), pp. 211–232] investigating changes in the ability to extract binaural information during temporal sequences of sounds.

THURSDAY MORNING, 5 DECEMBER 2013 POWELL, 10:00 A.M. TO 11:35 A.M.

## Session 4aSA

## Structural Acoustics and Vibration, Architectural Acoustics, and Noise: Human-Induced Vibration in Buildings

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

**Chair's Introduction—10:00**

### *Invited Papers*

10:05

**4aSA1. Prediction and measurement of floor response due to walking.** Thomas M. Murray (Civil Eng., Virginia Tech, 537 Wisteria Dr., Radford, VA 24141, thmurray@vt.edu) and Brad Davis (Civil Eng., Univ. of Kentucky, Lexington, KY)

The procedure for determine if a floor design will result in annoying vibrations due to walking in the American Institute of Steel Construction Design Guide 11 Floor Vibrations due to Human Activity is based on single degree of freedom response with a number of assumptions which reflect actual occupant conditions. Understanding the development of the procedure is necessary when comparing the results of field testing of problem floors. Development of the procedure, its accuracy when compared to a large data base of tested floors, and comments on appropriate testing protocols are presented.



10:25

**4aSA2. Floor vibration testing in hospital operating rooms.** Anthony Nash (Charles M. Salter Associates, 130 Sutter St., Ste. 500, San Francisco, CA 94104, anthony.nash@cmsalter.com)

Operating rooms in several hospitals have been studied for their *in-situ* floor vibration properties. The motivations ranged from identifying the cause of unstable images observed in a surgical microscope to determining whether a portion of a long span floor in a hospital parking garage could be adapted for a future surgical center. As part of these studies, several types of floor excitation techniques were employed including a vibration exciter, a "heeldrop," and a human walker. The input magnitudes from the first two of these excitation sources can be measured at a given location using a calibrated force plate. The human walker, however, is a "wild card" source since the location of the footfall is constantly changing; moreover, the nature of the applied force pulse is unknown under field conditions. Thus, one cannot easily quantify the dynamic force of the test signal that is most germane to the assessment of floor vibration in the field. The paper will review the literature governing floor vibration limits in sensitive buildings and summarize the utility of information obtained by various floor testing protocols.

10:45

**4aSA3. Improvement of footfall vibration of concrete floors: Two case studies.** Michael Gendreau and Hal Amick (Colin Gordon Associates, 150 North Hill Dr., Ste. 15, Brisbane, CA 94005, mgendreau@colingordon.com)

It is sometimes necessary to improve the footfall performance of an existing concrete floor in a manufacturing facility, as is the case when the vibration criterion changes because of new technology. The paper examines two case studies in which the floors were modified simply by placement of an additional concrete slab atop the existing slab. This approach has some drawbacks, most notably the additional weight, which must be accommodated by the available framing and foundation capacity. Measured data are presented which illustrate the changes in structural properties and footfall performance.

### *Contributed Papers*

11:05

**4aSA4. Investigation on floor impact noise difference between Rhamen and wall-column constructions using vibration analysis.** Dukyoung Hwang, Sinyoeb Lee, and Junhong Park (Hanyang Univ., Haengdang 1-dong, Seongdong-gu, Seoul 133-791, South Korea, dyhwang@hanyang.ac.kr)

Floor impact noise generation depends on the configuration of building structures. Especially, the Rhamen construction is known to radiate less than the wall-column construction. In this study, the sound radiation mechanisms between the two different structures are compared through experimentation using laboratory setup of the scaled model. The transfer of vibration energy from external impact source is calculated and compared with the measured results. Parameters affecting the radiated sound energy are determined and evaluated. The in and out-of phase vibrations of the building floors resulted in tonal sound radiation at two-closely located frequencies which resulted in modulated floor impact sound. This modulation increased the annoyance on the residents. Eventually, a design method to efficiently reduce the floor impact sound is proposed.

11:20

**4aSA6. Dissipative effects in the response of an elastic medium to a localized force.** Douglas Photiadis (NRL, 4555 Overlook Ave. SW, Washington, DC 20375, douglas.photiadis@nrl.navy.mil)

The effect of dissipation on the real part of the admittance of an elastic half-space is typically thought to be unimportant if the loss factor of the elastic medium is small. However, dissipation induces losses in the near field of the source and, provided the size of the source is small enough, this phenomenon can be more important than elastic wave radiation. Such losses give rise to a fundamental limit in the quality factor of an oscillator attached to a substrate. Near field losses associated with strains in the elastic substrate can actually be larger than intrinsic losses in the oscillator itself if the internal friction of the substrate is larger than the internal friction of the oscillator. [This research was sponsored by the Office of Naval Research.]

THURSDAY MORNING, 5 DECEMBER 2013 PLAZA B, 8:30 A.M. TO 12:00 NOON

## Session 4aSC

## Speech Communication: Speech Production I

Bryan Gick, Chair
*Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada*

### *Contributed Papers*

8:30

**4aSC1. A comparison of speech errors elicited by sentences and alternating repetitive tongue twisters.** Stefanie Shattuck-Hufnagel (Speech Commun., MIT, 36-511 MIT, 77 Mass Ave., Cambridge, MA 02139, sshuf@mit.edu), Cathy Bai (Wellesley College, Wellesley, MA), Mark Tiede (Haskins Labs., New Haven, CT), Argyro Katsikis (Univ. of Potsdam, Potsdam, Poland), Marianne Pouplier (Univ. of Munich, Munich, Germany), and Louis Goldstein (USC, Los Angeles, CA)

Sound-level errors collected by ear from continuous communicative speech have been interpreted as mis-selections of planning elements, which are then produced fluently without residue of the original target (Lashley 1957, Fromkin 1972, Garrett 1975, Shattuck-Hufnagel 1982). In contrast, articulatory measures of tongue twister errors reveal gestural intrusions: target and intrusion elements are co-produced, sometimes resulting in a gestural error which is imperceptible to listeners (Pouplier 2003, Goldstein *et al.* 2007; see also Mowrey and MacKay 1970). Is this apparent difference due to structure and processing differences between the two utterance types, i.e., sentences (e.g., The top cop saw a cop top) vs alternating repetitive word lists (e.g., top cop top cop top cop) generally produced with quasi-periodic timing? Or, do articulatory measures simply capture the nature of sound-level errors more accurately? We elicited errors using both types of stimuli in the same experimental session; perceptual and acoustic analyses show that sentences provoke more apparent whole-segment substitutions (e.g., /tap/ for /kap/), while alternating repetitive lists provoke more errors with two onset bursts (e.g., /tkap/), resembling gestural intrusions. This suggests that there may be more than one mechanism underlying spoken errors, and that different materials may engage these mechanisms to different degrees.

8:45

**4aSC2. An artificial neural network model for serial speech production and speech error simulation.** Erin Rusaw (Univ. of Southern California, 1150 Stanford St., Apt. 1, Santa Monica, CA 90403, ecrusaw@gmail.com)

This paper presents serial speech gesture encoding and recall extension to the Neural Oscillator Model Speech Timing and Rhythm (NOMSTR), previously used to simulate prosodic and syllable timing (Rusaw 2010, 2013), and describes how it can be used to simulate speech errors and

speech error patterns. This model uses NOMSTR to provide the phonological context vector which drives its serial recall, activating a series of gesture output nodes. Besides the prosodic context signal, the gesture output nodes have two other sources of input: (1) Nuclear gestures nodes provide excitatory input to the nodes required to produce the onset and coda gestures they share a syllable with; this mechanism is the ANN instantiation of the nucleus-centered syllable model. (2) Second, nuclear gesture nodes receive excitatory input from NOMSTR's syllable-level thresholded node, which ties the timing of the gestures in a syllable to the time at which the syllable occurs in the phonological output. This extended version of NOMSTR is shown to able to simulate aspects of speech error behavior which other models have been unable to explain, such as C/V error (O-C/N) asymmetry, and error dependence (Rusaw and Cole 2009). [Work supported by NSF and NIH.]

### 9:00

**4aSC3. Simulations of sound change resulting from a production-recovery loop.** Benjamin Parrell (Dept. of Linguist., Univ. of Southern California,GFS 301, Los Angeles, CA 90089, parrell@usc.edu), Adam Lammert (Signal Anal. & Interpretation Lab., Univ. of Southern California, Los Angles, CA), Shrikanth Narayanan (Signal Anal. & Interpretation Lab., Univ. of Southern California, Los Angeles, CA), and Louis Goldstein (Linguist., Univ. of Southern California, Los Angeles, CA)

We present a computational model of lenition-based sound change. Speech production targets for constriction degree are modeled by differential equations with a single stable fixed point at the target constriction degree that interact with higher order equations that reflect prosodically conditioned variation. This output is then input to the articulatory-to-acoustic forward map, the quantal nature of which causes continuous variation in constriction degree to result in two outcomes: closure or spirantization. The quantized acoustic outcomes are then used as the input for the language learner, who must recover the produced constriction degree in the ambient language environment to produce the speech unit appropriately. We model this recovery as estimating the parameters of an articulatory distribution that is most likely to have produced the acoustic observations (i.e., the maximum likelihood estimate). In particular, we optimize the parameters of a Gaussian distribution with respect to their log-likelihood function, given the acoustic data that results from applying the quantal forward map to that distribution. We show how this production-recovery loop may lead to sound change in a language by varying the nature of the articulatory-to-acoustic map as well as the amount of knowledge the learner has about both the prosodic structure and articulatory-to-acoustic map.

### 9:15

**4aSC4. Feedback-driven corrective movements in speech in the absence of altered feedback.** Caroline A. Niziolek, Srikantan S. Nagarajan, and John F. Houde (Univ. of California, San Francisco, 513 Parnassus Ave., Rm. S-362, Box 0628, San Francisco, CA 94118, cniziolek@ohns.ucsf.edu)

Altered auditory feedback often evokes a compensatory vocal response in speakers, providing evidence that errors in speech production can be rapidly corrected online. To assess whether the same compensatory mechanism is employed in natural, unaltered speech, we carried out an acoustic analysis characterizing formant movement during single speaking trials. Subjects produced 200 repetitions each of three different monosyllabic words in the MEG scanner and, separately, in the presence of varying background noise levels. To assess corrective responses, we compared the centricity of formant values at the beginning to that at the middle of each trial. In all subjects, we found strong evidence of vowel "centering"—that is, a corrective movement mid-utterance that caused utterances at the periphery to move closer to the center of the formant distribution. Across subjects, the magnitude of vowel centering was correlated with auditory cortical suppression preceding the corrective movement, suggesting that the suppression may serve as a neural mechanism for error detection and correction. These findings suggest that less-prototypical utterances, which make up a large proportion of natural speech, are processed as potential errors, and that feedback-driven speech error correction is occurring constantly on a small scale.

### 9:30

**4aSC5. Coarticulation as an epiphenomenon of syllable-synchronized target approximation—Evidence from fundamental frequency aligned formant trajectories in Mandarin.** Hong Gao (English Lang. and Lit., Sichuan Univ., Chengdu, China) and Yi Xu (Speech, Hearing and Phonetic Sci., Univ. College London, Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, yi.xu@ucl.ac.uk)

An experiment was carried out to test the hypothesis that the syllable is a time structure that synchronizes tonal, consonantal, and vocalic target approximation movements. The strategy was to align formant movements with F0 turning points of lexical tones as time reference, and then assess the temporal scope of articulatory movements by comparing formant trajectories and their turning points across minimal pairs. Native Mandarin speakers produced C1V1#C2V2 disyllabic sequences where C2 is /y/, /w/ or /l/, and V1 and V2 varied in height and frontness. Analysis of F0-aligned F2-3 (average of F2 and F3) trajectories revealed patterns in support of the main hypothesis. First, movements clearly discernable as approaching either C2 or V2 targets started at about the same time from the center of V1, i.e., well before the conventional landmark-based syllable boundary. Second, some F2-3 trajectories extended continuously from the center of V1 to the center of V2, across the intervening /l/, indicating a long and uninterrupted V2 approximation movement. These results provide support for the view that genuine CV co-production occurs only between onset C and the following V, while the rest of the "coarticulation" is only an epiphenomenon (arising from landmark-based segmentation) of syllable-synchronized target approximation.

### 9:45–10:00 General Discussion

### 10:00–10:30 Break

### 10:30

**4aSC6. Gradiency vs categoricity in the production of prosodic boundaries as reflected in different kinematic measures.** Jelena Krivokapic (Linguist., Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109-1220, jelenak@umich.edu), Christine Mooshammer (Institut für deutsche Sprache und Linguistik, Humboldt-Universität zu Berlin, Berlin, Germany), and Mark Tiede (Haskins Labs., New Haven, CT)

Two questions are examined in an EMA experiment: (1) are speech gestures at prosodic boundaries produced in a categorical or in a gradient manner (extending work by Krivokapic, Mooshammer, and Tiede 2013), and (2) how do different articulatory measures reflect boundary strength. Forty-eight sentences were constructed, each containing one, two, or three prosodic boundaries, for a total of 56 boundaries per sentence set. The range of predicted boundary types varied from a weak clitic boundary to a strong sentence boundary. Each boundary fell between the words "column and." Seven subjects read six to eight repetitions of these sentences. Various temporal properties of boundaries have been examined, including the duration of the lip opening movement for [m], which is the movement closest to the boundary (and therefore most strongly reflects the boundary properties), and the duration of the jaw movement from the opening movement for the first vowel in "column" to the opening movement for the postboundary vowel (a variable which spans the boundary). These measures have been subjected to Gaussian mixture model analysis. Results show evidence for a more fine-grained structure than can be predicted by the three prosodic levels of traditional models. [Work supported by NIH DC003172-16, DC008780, DC002717].

### 10:45

**4aSC7. Vowel locus equations as a measure of vowel coarticulatory aggressiveness.** Wei-rong Chen (Graduate Inst. of Linguist., National Tsing Hua Univ., 3029 S. Grand Ave., Apt. 10, Los Angeles, CA 90007, waitlong75@gmail.com) and Khalil Iskarous (Dept. of Linguist., Univ. of Southern California, Los Angeles, CA)

Studies in locus equations, a quantification of the degree to which F2 at vowel onset (or consonant place) can be predicted by F2 at vowel midpoint (or vowel place), have shown that the slope of locus equations is a reverse measure of coarticulatory resistance of consonants, in consonant-vowel (CV) sequences with C fixed and V varying. This study presents the first application of locus equations to the measure of coarticulatory properties of

4a THU. AM

vowels. The locus equations analysis for vowels, on our articulatory data (collected with electromagnetic articulograph, EMA) of CV syllables from 7 Taiwan Mandarin speakers, with V fixed and C varying, shows that vowel /i/ has the greatest slope among vowels at tongue body, whereas vowel /a/ and /u/ mostly do not distinguish from each other, and at the tongue tip, the slope does not vary across the three vowels /i/, /a/ and /u/. Based on the CV model theory of gestural coordination (C-V in-phase relation) and the formula of linear regression, we claim that the slope of vowel locus equations is positively related to coarticulatory aggressiveness of vowel, which is further supported by a comparison with a well-established measure of vowel coarticulatory aggressive, based on contextual variability, on the same data.

**11:00**

**4aSC8. An MRI comparison of /s/ production in four subject conditions.** Andrew D. Pedersen (Neural and Pain Sci., Univ. of Maryland Dental School, 650 W. Baltimore St Rm 8207, Baltimore, MD 21201, mstone@umaryland.edu), Jun Hwang (Orthodontics, Univ. of Maryland Dental School, Baltimore, MD), Jonghye Woo (Neural and Pain Sci., Univ. of Maryland Dental School, Baltimore, MD), Fangxu Xing, Jerry L. Prince (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), and Maureen Stone (Neural and Pain Sci., Univ. of Maryland Dental School, Baltimore, MD)

This study examined a control subject and three patients who had surgery to remove tongue cancer. One patient's surgery was closed with sutures, one with a radial forearm free flap reconstruction, and one with sutures plus radiation. This study aims to ascertain the effects of these traumas on internal and surface tongue motion during speech. The flap consists of soft tissue that is vascularized, but not innervated; it increases tongue bulk, but has no direct motor control. The other two patients have missing tissue and a scar where the cut regions were sewn together. This morphological change may increase difficulty creating properly formed palatal contacts. The supplemental radiation treatment may cause additional muscle stiffness due to fibrosis. The cine and tagged datasets were recorded in axial, coronal, and sagittal orientations using identical parameters so their data could be overlaid. Each dataset was reconstructed into 3D volumes, one for each time-frame in the word. From each cine-MRI volume, the 3D tongue surface was segmented and used as a "mask" in the tagged-MRI volume. In the tagged-MRI volumes, 3D displacement fields were calculated to show motion of each tissue point inside the tongue mask during the speech task.

**11:15**

**4aSC9. Visualization of the laryngeal muscles with high-quality magnetic resonance imaging.** Sayoko Takano (Media Information Sci., Kanazawa Inst. of Technol., 7-1 Ohgigaoka, Nonoichi 921 - 8501, Japan, tsayoko@neptune.kanazawa-it.ac.jp) and Kiyoshi Honda (Tianjin Univ., Tianjin, China)

The larynx is the organ to adjust voice in speech important for controlling voice phonation, and the tensions of the vocal folds cord is controlled by the relative positions among the thyroid, arytenoid, and cricoid cartilages. Magnetic resonance images (MRI) have been employed to observe the positions of those cartilages, while however, visualizing the laryngeal muscles for activating moving those cartilages is still difficult with the conventional MRI. This study aims to visualize and identify the laryngeal muscles based on the composition of the three different sets of gray-scale MRI data into red, green, and blue channels, namely RGB-MRI. The types of imaging parameters image variations are field-echo opposed-phase (FE(op)), spin-echo proton density weighted (SE(PDw)), and spin-echo T1 weighted (SE(T1w)). This method for image composition helps us identifying could reveal the location of the muscles in the laryngeal region *in vivo*. We have successfully identified the sternohyoid muscle, thyrohyoid muscle, lateral thyroarytenoid muscle, cricothyroid muscle, stenrocleidmustoid muscle, omohyoid muscle, and inferior constrictor muscles. The RGB-MRI could give further provide anatomical information of the larynx, especially about the muscle location and its deformation involved in on speech mechanisms, which will be useful for both research experts and beginners.

**11:30**

**4aSC10. Motor control primitives arising from a dynamical systems model of vocal tract articulation.** Vikram Ramanarayanan, Louis Goldstein, and Shrikanth Narayanan (Univ. of Southern California, 3740 McClintock Ave., EEB421, Los Angeles, CA 90089-2564, vramanar@usc.edu)

We have previously presented a computational approach to derive interpretable movement primitives from speech articulation data using a convolutive Nonnegative Matrix Factorization with sparseness constraints (cNMFsc) technique (Ramanarayanan *et al.*, Interspeech 2011; Ramanarayanan *et al.*, J. Acoust. Soc. Am. **134**(2), in press). However, it is not clear whether finding such a dictionary of primitives can be useful for speech motor control, particularly in finding a low-dimensional subspace for such control. In this paper, we examine this possibility in two steps. First, we use the iterative Linear Quadratic Gaussian (iLQG) algorithm to derive a set of control inputs to a dynamical systems model of the vocal tract that produces a desired movement sequence. Second, we use the cNMFsc algorithm to find a small dictionary of control input ''primitives'' that can be used to drive said dynamical systems model of the vocal tract to produce the desired range of articulatory movement. We show, using both qualitative and quantitative evaluation on synthetic data produced by an articulatory synthesizer, that such a method can be used to derive a small number of control primitives that produce linguistically interpretable and ecologically valid movements. Such a primitives-based framework could help inform theories of speech motor control and coordination.

**11:45–12:00 General Discussion**

# Session 4aSP

## Signal Processing in Acoustics: Alternative Array Spacing and Time Sampling Techniques

Andrew T. Pyzdek, Cochair
*Appl. Res. Lab., The Penn State Univ., PO Box 30, State College, PA 16804*

Jeffrey A. Ballard, Cochair
*Appl. Res. Labs., The Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029*

### *Invited Papers*

**9:55**

**4aSP1. Detection performance of coprime sensor arrays.** Kaushallya Adhikari and John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts, Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, kadhikari@umassd.edu)

Coprime sensor arrays (CSAs) achieve the resolution of a fully populated uniform linear array (ULA) with the same aperture using fewer sensors. The CSA's reduced number of sensors diminishes its ability to attenuate white noise, and consequently also reduces its array gain. Assuming narrowband independent signal and white noise, when the signal and the noise are modeled by circular complex zero mean Gaussian distributions, the detection statistic distribution for the ULA simplifies to an exponential distribution with its parameter dependent on the input SNR. The detection statistic for the CSA is the product of the output of one subarray with the complex conjugate of the output of the second subarray. Being the product of two independent complex Gaussian random variables, the CSA detection statistic has a distribution proportional to a modified zeroth order Bessel function of the second kind [O'Donoughue and Moura, IEEE Signal Process. (2012)]. Manipulating the ULA and CSA detection statistic distributions provides analytical expressions for probabilities of detection and false alarm and also mean discriminating information. Evaluating these expressions confirms that a CSA pays a detection gain penalty of about 5 dB over a wide range of SNRs and array sizes. [Work supported by ONR.]

**10:15**

**4aSP2. The application of compressive sensing to underwater acoustic array processing and design.** Jeffrey S. Rogers, Charles F. Gaumond, and Geoffrey F. Edelmann (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375, jeff.rogers@nrl.navy.mil)

A brief overview of the application of compressive sensing to underwater acoustic array processing is given. Specific algorithms for estimating direction of arrival via L1 minimization are introduced. The Statistical Reduced Isometry Property (StRIP) is defined and used as a method to determine the ideal array designs for compressive beamforming. The tradeoff of using nested apertures and non-integer element spacing for improving StRIP over a broad range of frequencies is studied numerically. The array gain, which is a more directly useful performance metric, is also estimated using Monte Carlo methods. Compressive beamforming results from at-sea data taken on the Five Octave Research Array (FORA) will be presented. [Research funded by the Office of Naval Research.]

**10:35**

**4aSP3. Coprime arrays in the context of compressive sensing.** Andrew T. Pyzdek and R. Lee Culver (Appl. Res. Lab., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu)

Arising naturally from the formalism of coarrays, coprime arrays offer a means of achieving sparsity along an array while maintaining certain measures of performance. This configuration offers benefits over traditional configurations for large aperture towed arrays in the ocean environment, but shallow water propagation complicates the wide sense stationarity assumption that is essential to proper application of coarray concepts. Compressive sensing techniques allow for the reconstruction of generalized signals which are sparse on some known basis. Existing literature has shown the connection between compressive sensing and random arrays, drawing direct parallels between the results and methods that arose independently in both fields. Compressive sensing may offer an alternative understanding of coprime arrays, which is independent of wide sense stationarity, and thus applicable to shallow water environments. Coprime arrays will be considered in the context of compressive sensing. It will be discussed if the measurement matrices that arise from this array geometry might possess properties necessary for compressive sensing reconstruction to function, such as the Restricted Isometry Property. [This research was supported by the Applied Research Laboratory, at the Pennsylvania State University through the Eric Walker Graduate Assistantship Program.]

## *Contributed Papers*

**10:55**

**4aSP4. Near-field beamforming with augmentable non-uniform arrays for far-field interference suppression.** Jonathan Odom and Jeffrey Krolik (Elec. and Comput. Eng., Duke Univ., PO Box 90291, Durham, NC 27708, jonathan.odom@duke.edu)

This paper addresses the problem of adaptive near-field beamforming in the presence of many far-field interferers. Minimally redundant and, more generally, augmentable non-uniform arrays span larger array apertures and permit adaptive discrimination of more sources than sensors but require spatial-stationarity and increased snapshot support, which precludes their use for dynamic, near-field sources. In this paper, adaptive beamforming with augmentable non-uniform arrays is proposed which nulls relatively static far-field interference while enabling the detection of dynamic near-field sources. The use of an augmentable under-sampled array permits formation of a covariance matrix that can be used to implement reduced-rank dominant rejection of more interferers than sensors. Meanwhile, near-field beamforming in bearing and range is achieved using only the physical non-uniformly spaced sensor locations. An interference dominated environment is simulated to demonstrate the increase of array gain achieved over current techniques in the low snapshot support regime. [Work sponsored by ONR.]

**11:10**

**4aSP5. Coprime microphone arrays for direction-of-arrival estimation.** John P. Nichols and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, jpnichols9@gmail.com)

Direction-of-arrival estimation using microphone arrays is relatively simple with low signal-to-noise ratio, but precise location estimation amongst noise requires many sensors to reduce beam width. Coprime linear microphone arrays allow for narrow beams with fewer sensors. A coprime microphone array is made up of two overlapping uniform linear arrays with $M$ and $N$ sensors, where $M$ and $N$ are coprime. By applying spatial filtering with both arrays and combining their outputs, $M + N$ sensors can yield $MN$ directional bands. In this work, a coprime microphone array is used to estimate the location of multiple uncorrelated narrowband sources hidden in noise.

**11:25**

**4aSP6. Beam-patterns of a collocated unit of three orthogonally oriented second-order directional sensors.** Kainam T. Wong and Yang Song (Dept. of Electron. & Information Eng., Hong Kong Polytechnic Univ., DE 605, Hung Hom KLN, Hong Kong, ktwong@ieee.org)

A microphone or a hydrophone measures the incident acoustic wavefield as a scalar field of pressure. Underlying this pressure scalar is the particle-velocity vector, which constitutes the spatial gradient of the wavefield. Each Cartesian component of this gradient vector may be directly measured (without computing any spatial derivative) by a particle-velocity sensor aligned in parallel to that Cartesian coordinate. The entire $3 \times 1$ gradient vector may be measured at any point in space, using three such particle-velocity sensors, collocated but orthogonally oriented among themselves. Such a collocated triad has an array manifold independent of signal frequency, thus uncoupling azimuth-elevation beamforming from the frequency-dimension. This triad's beam-pattern has already been investigated by Wong and Chi ["Beam patterns of an underwater acoustic vector hydrophone located away from any reflecting boundary," IEEE J. Ocean. Eng. **27**(3), 628–637 (2002)]. While the particle-velocity sensor measures the first-order spatial derivative of the pressure field, a second-order spatial derivative of the pressure-field could likewise be defined and be measured by a "second-order directional sensor" that would have sharper directivity. This paper follows up that investigation, but now for second-order directional sensors.

**11:40**

**4aSP7. Objective functions incorporating various norms for the three-dimensional sound manipulation.** Yang-Hann Kim (Ctr. for Noise and Vib. Control, KAIST, Dept. of M.E., Sci. Town, Daejon-shi 305-703, South Korea, yanghann@kaist.ac.kr), Jung-Woo Choi, and Min-Ho Song (Ctr. for Noise and Vib. Control, KAIST, Daejeon, South Korea)

Using many array speakers to manipulate sound in space and time requires proper objective functions that can determine the array speakers' magnitude and phase relationships. Mathematically speaking, this means that magnitude and phase of each speaker have to be determined in such a way that certain designed objective functions can be minimized or maximized. The most popular way to define the objective function is to utilize the form of L2 norm, but it is also likely possible to utilize L1 norm to decide corresponding array speakers magnitudes and phases. Depending on what we would like to manipulate in space and time, the objective functions can be selected: they can be, for example, energy in a selected zone, or contrast of sound energy, for the case of L2 norm. In this paper, we study how those functions affect the performance of sound manipulation in terms of array parameters and effectiveness of 3D sound manipulation.

**11:55**

**4aSP8. Modeling imaging performance of multistatic acoustic arrays of non-uniform geometries.** Michael Lee, Michael Liebling, and Hua Lee (Elec. and Comput. Eng., Univ. of California, Santa Barbara, Harold Frank Hall, Rm. 4155, Santa Barbara, CA 93106-9560, michaellee@umail.ucsb.edu)

Unlike the most arrays in acoustical imaging systems, a system based on a dynamically reconfigurable multistatic array would be able to physically adapt to the volume of interest, improving angular coverage of the illuminating array aperture and thus resolution of target images. With such arbitrary distributions of array elements, the extent to which each element contributes to the image reconstruction of targets may change with each imaging cycle. For this reason, the acoustic beamspread and direction of each array element become dominant variables in the resolving capability and data processing of the system. In this paper, non-linear element distributions with specified beamspreads are simulated to image point targets for the purpose of analyzing this relationship and its relevance to spatial bandwidth. MATLAB simulations based on coherent illumination and image reconstruction provide the foundation for this work. By adding the capability to specify beamspread angles and direction for each element, imaging simulation results have more practical value. For any given array configuration, the region of interest can now be visualized in terms of favorability for acoustic illumination. Likewise, the beamspread specifications enable the image reconstruction algorithm to more reliably reconstruct targets.

# Session 4aUW

## Underwater Acoustics and Acoustical Oceanography: The Acoustics of Bubbles and Bubble Clouds in the Ocean

Grant B. Deane, Cochair
*Marine Physical Lab., Univ. of California, San Diego, La Jolla, CA 92093-0238*

R. Lee Culver, Cochair
*ARL, Penn State Univ., PO Box 30, State College, PA 16804*

**Chair's Introduction—8:55**

### *Invited Papers*

**9:00**

**4aUW1. What is the surface tension at a bubble wall?** Tim Leighton, Mengyang Zhu (Inst. of Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton, Hampshire SO17 1BJ, United Kingdom, tgl@soton.ac.uk), and Peter Birkin (School of Chemistry, Univ. of Southampton, Southampton, United Kingdom)

The surface tension on a bubble wall frequently arises in models of single bubbles and is an influential parameter when assessing many important processes. These range from the global transfer of mass, momentum, and energy between the oceans and the atmosphere, to the efficacy of microbubbles for clinical diagnosis and therapy. However, the simplicity by which surface tension appears in formulations belies the complexity of the process. For example, the distribution of chemical species or dopants may be uneven over the wall of a single bubble, the translation of a bubble relative to the bulk liquid may change the chemical loading over time, and the pulsation and shape changes of bubbles perturb the distribution of a dopant on the bubble wall. This paper reports on an experimental technique for estimating one measure of the surface tension at the gas/liquid boundary of a bubble.

**9:20**

**4aUW2. The influence of temperature on bubble formation under breaking waves.** Helen Czerski (Inst. for Sound and Vib. Res., Univ. of Southampton, Highfield, Southampton SO17 1BJ, United Kingdom, h.czerski@soton.ac.uk)

The bubble plumes generated by breaking waves have a complex structure that is highly dependent on the local environment. These temporary bubble populations have a strong influence on many processes at the air-sea interface, for example, air-sea gas transfer, aerosol production, sound transmission, and optical absorption. To quantify the importance of the bubble plume, two things are required: the state of the initial bubble plume and an understanding of the longer-term plume evolution. This paper focuses on the formation of the initial bubble plume. I'll present evidence showing the effect of temperature on the fragmentation of single bubbles in the laboratory. These results imply that changing the water temperature has consequences for ocean bubble plume structure, bubble size distribution, and whole bubble plume acoustics, and these will be discussed.



### *Contributed Papers*

**9:40**

**4aUW3. Acoustic properties of oil/gas/sea water mixtures.** R. Lee Culver (ARL, Penn State Univ., PO Box 30, State College, PA 16804, rlc5@psu.edu)

This paper presents an approach to calculating the acoustic properties, i.e., sound speed, and scattering cross-section, of oil, oil/gas, and oil/gas/sea water mixtures. The petroleum industry characterizes reservoir fluid in terms of the relationship between pressure, volume and temperature (PVT) of the oil utilizing the formation volume factor (FVF), which is defined as the ratio of the volume of oil at a pressure and temperature relative to a stock tank barrel (STB) at surface conditions. Another important oil reservoir parameter is the gas-oil ratio (GOR), which is the percent by volume of gas which is present in the oil at reservoir conditions (the gas is typically entirely dissolved—i.e., none is free—due to the great pressure). The reservoir fluid is typically under-saturated, meaning more gas could be dissolved in the oil, were it present. As the oil rises to the surface, the pressure and temperature drop. Once they drop below the bubble point, gas begins to come out of solution. Also, if fluids (oil and gas) are released into the ocean, sea water will become mixed in with the well fluids. The interest here is in the acoustic properties of oil only, oil/gas and oil/gas/sea water mixtures. The approach is to calculate the density and compressibility of the constituents and from them predict the aggregate acoustic properties using mixture theory.

**9:55**

**4aUW4. Underwater sound radiated by bubbles released by melting glacier ice.** Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Erin C. Pettit (Dept. of Glaciology and Geophys., The Univ. of Alaska Fairbanks, Fairbanks, AK)

Passive acoustics monitoring techniques have been examined as a method to remotely sense activity of glacier ice near the ice-ocean boundary [Ann. Glaciol. **53**, 113–121 (2012)]. Sound from glacier calving events and the resultant breaking and bobbing of the ice after impact with the water ranges from infrasound (<10 Hz), to low-frequency (100 Hz—1 kHz), to higher frequency sound (>10 kHz) generated by breaking tsunamis and seiches after impact. Bubbles are known to form within ice during glacier formation and can be released from glaciers as they undergo submarine melting. Due to the possibility of high internal bubble pressure, this release can occur in the form of jetting or squirting events. Signals hypothesized to be from bubbles being released from melting glacier ice were measured in Unakwik Inlet and Icy Bay, Alaska using passive autonomous hydrophone moorings and near-surface recordings in the 1 kHz—3 kHz frequency range. To temporally and spatially correlate such acoustic emissions with bubble activity, a set of laboratory measurements was performed using small samples of glacier ice, and acoustic emission was positively correlated with bubble release. Taken together, these measurements support the use of passive acoustics to monitor marine glacier ice melt.

**10:10–10:30 Break**

**10:30**

**4aUW5. Low scatterer density limit of effective medium theory.** Craig N. Dolder (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu), Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Often effective medium theories assume that there are multiple scatterers per wavelength; however, practical limits for these models can be far more flexible. Commander and Prosperetti's model for sound propagation in bubbly liquids assumes that "an averaging volume must contain many bubbles" but does not explicitly assume that there must be many bubbles per wavelength, though that assumption might be considered implicit in the use of effective medium theory. The results of Foldy's exact multiple scattering theory are compared to the effective medium theory for very low scatterer densities in order to determine the low scatterer limit for which the effective medium theory holds. This is compared with experimental results with low scatterer densities. It is shown that effective medium models for bubbly liquids hold even when there are as little as one scatterer per wavelength. [This work was supported by the Office of Naval Research.]

**10:45**

**4aUW6. Time and frequency analysis of air bubbles injected underwater.** Hannan Lohrasbipeydeh (Dept. of Elec. Eng., Univ. of Victoria, EOW 448, 3800 Finnerty Rd., Victoria, B.C. V8P 5C2 Canada, lohrasbi@uvic.ca), Tom Dakin (Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and T. Aaron Gulliver (Elec. Eng., Univ. of Victoria, Victoria, BC, Canada)

The acoustic signal generated by vibrating air bubbles injected in water allows for the passive detection of underwater sources such as divers and gas leaking from pipes. In this paper, an experimental analysis of the time and frequency characteristics of these signals is presented. The decaying acoustic signal of a single bubble created by nozzles of different diameters is described and extended to a train of bubbles from an array of nozzles. The oscillatory motion of the bubble surface generates an acoustic signal that has a damped sinusoidal characteristic. The spectrum of this waveform has a fundamental frequency and several harmonics. Therefore, the effect of ambient noise can be suppressed using a suitable band pass filter with a center frequency equal to the bubble fundamental frequency. It is shown that the acoustic signal generated by these bubbles can be utilized for passive detection and remote sensing (including depth estimation). Experimental data from a Saanich inlet experiment is analyzed to obtain realistic results. These results are compared with the theoretical analysis.

**11:00**

**4aUW7. The limits of using closed-cell foam as a pressure release condition when dealing with underwater acoustics.** Craig N. Dolder and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu)

Closed-cell foams are frequently used in both the research and industry communities to provide a pressure release condition for underwater sound. In practice, closed-cell foams can effectively imitate a pressure release condition even if the thickness of the foam is as small as 1/60th the wavelength of the sound in water, however these conditions start to break down as the sound speed in water is reduced. The presence of even a small volume fraction of bubbles can cause the sound speed to drop in such a way that the water can couple with the closed-cell foam and prevent the interface from providing a pressure release condition. Measurements made in an acoustic resonator are used to show how the idealized system breaks down. [This work was supported by the Office of Naval Research.]

## Session 4pAA

# Architectural Acoustics: The Enduring Contributions of Two Giants in Building Acoustics: Ronald L. McKay and Warren E. Blazier

David A. Conant, Cochair
*MCH Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362*

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

**Chair's Introduction—1:00**

### *Invited Papers*

**1:05**

**4pAA1. Ronald L. McKay, FASA: The Los Angeles years.** David A. Conant (McKay Conant Hoover Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, dconant@MCHinc.com)

Ron McKay already had two very productive decades in various BBN field offices by the time we met in 1977, and was a high-level "generalist" with a growing portfolio of respected work in performing arts facilities. After 10 years together at BBN's Los Angeles office, we established our own consulting practice, McKay Conant Brook inc, with an emphasis on spaces for large public assembly. This paper focuses on Ron's contributions in the area of his special passion—performing arts—and various acknowledgments and awards leading to the prestigious AIA National Honor Award for Collaborative Achievement in 1999. His finest work was yet to come.

**1:25**

**4pAA2. Growth in acoustics consulting at Bolt, Beranek and Newman with Ron and Warren.** Carl Rosenberg (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, crosenberg@acentech.com), William J. Cavanuagh (Cavanaugh Tocci, Natick, MA), and Eric Wood (Acentech Inc., Cambridge, MA)

Acoustics consulting was a growing field in technology support to the architectural world in the second half of the 20th century, spearheaded by Bolt Beranek and Newman and other new firms. As part of this growth, BBN embarked on a program of establishing branch offices around the country. Ron McKay and Warren Blazier were outstanding technical leaders in their respective fields and also vanguards of these developments, although both found fulfillment later in their careers away from BBN. This paper shares the growth and evolution of the profession through these two paragons of consulting leadership, and acknowledges the profound contributions of their early years to the reputation of the acoustics consulting world.

**1:45**

**4pAA3. On the shoulders of giants: Remembering the contributions of Warren Blazier.** Robert F. Mahoney (Robert F Mahoney & Assoc., 310 Balsam Ave., Boulder, CO 80304-3238, rfm@rfma.com)

Gentleman, scholar, mentor, Warren Blazier exemplified all these roles in his career of six decades. Many of us who benefited from his research, his teaching, and especially from his unstinting and magnanimous instruction owe Warren a great deal. This brief talk will discuss some of the frontiers Warren established—and significantly advanced—as well as the lessons we beneficiaries learned from him. Many of these lessons extend well beyond the science of noise control and, we hope, can be passed on to our successors as well.

**2:05**

**4pAA4. Warren Blazier's contributions to building acoustics, a manufacturer's perspective.** Norman Mason (Mason Industries, Inc., 350 Rabro Dr., Hauppauge, NY 11788, nmason@mason-ind.com)

Warren Blazier and Norm Mason first met at York Borg-Warner's Engineering Offices over 50 years ago; Warren was running their Acoustics Department. Warren moved on to Bolt Beranek Newman and then physically to San Francisco and the two remained great friends. Norm Mason consulted Warren on his theoretical understanding of Noise and Vibration Control and Warren was influential in the development of Mason Industries Architectural products. This paper will attempt to capture the highlights of Warren Blazier's friendship with Mason Industries, Inc.

**2:25–2:40 Break**

**2:40**

**4pAA5. Lessons from the dean of noise control.** Jim X. Borzym (Borzym Acoust. LLC, 2221 Columbine Ave., Boulder, CO 80302, acoustics@columbine.net)

Professorial, collegial, passionate, and gentlemanly, Warren E. Blazier, Jr., was admired in our profession as a top-level consultant and collaborator, and sometimes referred to as the "Dean" of noise control engineering. This presentation will highlight several technical topics incorporated in project consultations by Warren Blazier. Topics include duct plena, fan efficiency, structural resonance, floated floors, and unusual projects. Comments and personal insights presented by a mentee and collaborator of Warren Blazer's during the post-BBN decade 1986 through 1996.

**3:00**

**4pAA6. Warren Blazier—The consultant's consultant.** Richard Talaske (TALASKE | Sound Thinking, 1033 South Bldg., Oak Park, IL 60302, rick@talaske.com)

Many acoustic consultants turned to Warren for advice or confirmation of design solutions to solve their toughest noise control design challenges, this consultant included. Spanning nearly three decades, Warren collaborated with TALASKE | Sound Thinking technically as Associated Acoustical Consultant on retainer and business-wise as a member of the Board of Directors. Insights into the expertise and personality of Warren will be provided, with added contributions by Jerry Hyde.

**3:20**

**4pAA7. Warren Blazier and his contributions to noise rating criteria.** Kenneth P. Roy (Bldg. Products Technol. Lab, Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17603, kproy@armstrong.com)

Warren Blazier was well known for his work with sound and vibration in mechanical systems, e.g., HVAC equipment. As such, he was particularly active at ASHRAE in addition to the ASA meetings. Although I was always focused more on the "room acoustics" side of the design, I none-the-less offered to help edit the 1999 ASHRAE Handbook chapter 43 with Warren and Chuck Ebbing. This section dealt with "Indoor Sound Criteria," and what an eye opening event this turned out to be. You might say that I jumped into the fire as there was an ongoing "tug-of-war" over which rating system should be adopted to describe background noise in buildings from mechanical systems. NC, RC, RC mark II, NCB … several of these had been used up to this point and lots of controversy existed as to what to include in the re-write. A somewhat historic special meeting of ASHRAE TC2.6 was held in Boston on the evening of Sunday 27 June 1997 from 3:30 to 6:30 pm where this was discussed to resolve the issue. Both Warren and Leo Beranek were asked to make presentations on the merits of the competing approaches. And this is what was decided!!

**3:40**

**4pAA8. Deep roots and spreading branches—A shared legacy of acoustic DNA.** Larry Kirkegaard (Kirkegaard Associates, 801West Adams St., Chicago, IL 60607, lkirkegaard@kirkegaard.com) and Len Auerbach (Auerbach Pollock Friedlander, New York, NY)

Ron McKay and Warren Blazier were part of a generation of strong would-be consultants that were drawn to Bolt Beranek and Newman in the early to mid-sixties. They brought with them broad skills, great curiosity, and a contagious spirit of collaboration. Harvard and MIT were prime sources of talent, but BBN was magnetic to talent as well as project work from around the world. To know and appreciate Ron and Warren, you must know the richness of talent that surrounded and influenced them—their professional companions. Imagine a lively discussion with input from the likes of Leo Beranek, Dick Bolt, Bob Newman, Bill Cavanaugh, Ted Schultz, David Klepper, Bill Watters, Rein Pirn, George Kamperman, Bob Hoover, Carl Rosenberg, Layman Miller, Jacek Figwer, Jack Curtis, Russell Johnson, Bob Wolff, Tom DeGaetani, Len Auerbach, Dennis Paoletti, Joel Lewitz, and Dave Conant among many others. Arm-to-arm they could reach around the world. Their legacies enrich our profession in a myriad of ways. This paper shares memories and insights into both the genius and humanness of an important generation of our colleagues.

## Session 4pAB

## Animal Bioacoustics and Noise: Bioacoustic Contributions to Soundscapes II

John Hildebrand, Cochair
*Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093*

Simone Baumann-Pickering, Cochair
*Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093*

### *Invited Papers*

**1:30**

**4pAB1. Cyclical patterns in long-term bioacoustic data.** Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu), Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA), Ana Širović, and John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Cyclical patterns in biological systems occur from small to large temporal scales, and these cycles are also reflected in acoustic data. To interpret long-term patterns appropriately, knowledge about natural cycles is crucial. We will show examples of diel, lunar, and seasonal patterns for a variety of species from invertebrates to vertebrates. The observed patterns are likely driven by intra-specific communication, or prey behavior and availability, which in turn can be related to large scale and long-term environmental modulation. Absence of acoustic signals does not necessarily indicate absence of the caller but the calling behavior may be limited to a certain time of the cycle period. In addition, various acoustical sources can complicate results. Patterns from abiotic and anthropogenic sources may mask or alter natural biological acoustic cycles. Animals moving just outside of an acoustic recorder's detection range may lead to misinterpretation of diel behavior. Effects of multi-annual or multi-decadal cycles are extremely difficult to investigate due to the long time series needed to take them into consideration.

**1:50**

**4pAB2. Patterns in bioacoustic activity observed in U. S. National Parks.** Megan F. McKenna, Dan J. Mennitt, Emma Lynch, Damon Joyce, and Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO CO, megan_f_mckenna@nps.gov)

The Natural Sounds and Night Skies Division of the U.S. National Park Service has collected month-long acoustic recordings at more than 300 sites in 73 park units located throughout the United States, dating back to 2000. Each monitoring session lasted 25 days or more; some sites were monitored more than once. At all sites a calibrated Sound Level Meter recorded acoustic data in one-second, one-third octave band resolution; at many sites, simultaneous continuous acoustic recordings were collected using a digital audio recorder. These data were analyzed to identify broad patterns in bioacoustic activity within one-third octave bands. These bioacoustical patterns were analyzed in relation to site characteristics, seasons, and anthropogenic noise levels to identify significant associations. The resultant model could be used to produce a map predicting bioacoustical activity throughout the coterminous United States.

**2:10**

**4pAB3. Baleen whale calls in the Southern California Bight from 2009 to 2012.** Ana Sirovic (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., MC 0205, La Jolla, CA 92093-0205, asirovic@ucsd.edu), Marie A. Roch (San Diego State Univ., San Diego, CA), Simone Baumann-Pickering, Jasmine Buccowich, Amanda Debich, Sarah C. Johnson, Sara M. Kerosky, Lauren K. Roche, Sean M. Wiggins, and John A. Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA)

Baleen whales are an important contributor to the Southern California Bight soundscape. Calls from some species only contribute seasonally, while others are part of the soundscape year-round, albeit at varying levels. Passive acoustic monitoring has been conducted at the U.S. Navy's Southern California Offshore Range (SCORE) since 2009. Data were collected at two sites concurrently using High-frequency Acoustic Recording Packages (HARPs). Analysis in the low frequency band (10 Hz—1 kHz) has yielded results on the seasonal and interannual variation in the presence of calling blue (*Balaenoptera musculus*), fin (*B. physalus*), Bryde's (*B. edeni*), and humpback whales (*Megaptera novaeangliae*). Calls of most species (blue, Bryde's, and humpback whales) were only present during part of the year, indicating seasonal migration common for these species. On the contrary, fin whale calls were prevalent year-round, although the abundance of their 20 Hz calls tended to decrease in the summer. The link between the relative changes in interannual call abundance (i.e., soundscape contribution) and prevailing environmental conditions, such as the sea surface temperature and chlorophyll a, was investigated using the Tethys spatial-temporal database framework. These links can be important for understanding the year-to-year variation in soundscape.

## *Contributed Papers*

**2:30**

**4pAB4. Deep-diving cetaceans and the Deepwater Horizon oil spill.** Karlina Merkens (UC San Diego, SIO, 9500 Gilman Dr., MC 0205, La Jolla, CA 92093-0205, kmerkens@ucsd.edu), Mark McDonald (Whale Acoust., Bellvue, CO), Simone Baumann-Pickering, Kaitlin Frasier, Sean Wiggins, and Hildebrand John (UC San Diego, SIO, La Jolla, CA)

The Gulf of Mexico is home to at least six species of deep-diving cetaceans, including beaked whales, sperm whales, and dwarf and pygmy sperm whales. These species are all found in the region that was impacted by the Deepwater Horizon oil spill. Using High-frequency Acoustic Recording Packages (HARPs), we monitored for their presence at three deep-water sites. From over two years of wideband (10 Hz—100 kHz) recordings, the detections of deep-diving cetacean sounds were related to environmental and anthropogenic factors using Generalized Additive Models to identify relevant features. The modeling showed that the significance of habitat parameters varies by species and site, although lunar illumination and sea surface height anomaly were significant for most species at all sites. The relationships between the acoustic presence of the cetaceans and their environment help provide an understanding of the ecology of these species as well as the potential impact of the oil spill on their habitat. This material is based upon work supported by BP and NOAA under Award Number 20105138. Any opinions, findings, and conclusions or recommendations expressed in this publication are those of the author(s) and do not necessarily reflect the views of the BP and/or any State or Federal Natural Resource Trustee.

**2:45**

**4pAB5. Prospects for short-term and long-term passive acoustic monitoring of environmental change impact on marine mammals.** Natalia Sidorovskaia (Phys., Univ. of Louisiana at Lafayette, UL BOX 44210, Lafayette, LA 70504-4210, nas@louisiana.edu), Azmy S. Ackleh (Mathematics, Univ. of Louisiana at Lafayette, Lafayette, LA), Christopher O. Tiemann (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), Juliette W. Ioup, and George E. Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

The Littoral Acoustic Demonstration Center, LADC, a consortium of scientists from four Gulf state universities and the U.S. Navy, was begun in 2001 to study underwater noise and acoustic propagation, and the impact of human activities in the ocean on marine mammals, with emphasis on the Gulf of Mexico (GoM) region. LADC has a library of broadband passive acoustic data, collected by autonomous bottom-moored buoys, which sampled the GOM region ambient noise state, seismic airgun array emissions, and/or marine mammal activities six times during the last decade. LADC acoustic data represent an opportunity to study the short- and long-term effects of environmental changes on the marine mammal population. Environmental factors include baseline anthropogenic noise levels, passages of tropical storms, seismic exploration surveys in the area, and the 2010 Deepwater Horizon oil spill accident. The talk summarizes recent findings on the relationship between regional population dynamics of sperm and beaked whales and abrupt environmental changes with emphasis on the recent GoM oil spill. Statistically significant results of the study suggest a need for establishing consistent acoustic monitoring protocols in the oceanic areas of current or potential industrial activities. [Past data acquisitions were supported by ONR, SPAWAR, JIP, NSF, and Greenpeace.]

**3:00**

**4pAB6. Tethys: A workbench for bioacoustic measurements and environmental data.** Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., 5500 Campanile Dr., San Diego, CA 92182-7720, marie.roch@sdsu.edu), Simone Baumann-Pickering, Daniel Hwang, Heidi Batchelor (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Catherine L. Berchok (Fisheries Sci. Centers, NOAA, Seattle, Washington), Danielle Cholewiak (Fisheries Sci. Centers, NOAA, Woods Hole, Massachusetts), John A. Hildebrand (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Lisa M. Munger, Erin M. Oleson (Fisheries Sci. Centers, NOAA, Honolulu, Hawaii), Shannon Rankin (Fisheries Sci. Centers, NOAA, San Diego, California), Denise Risch (Fisheries Sci. Centers, NOAA, Woods Hole, Massachusetts), Ana Širović (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Melissa S. Soldevilla (Fisheries Sci. Centers, NOAA, Miami, Florida), and Sofie M. Van Parijs (Fisheries Sci. Centers, NOAA, Woods Hole, Massachusetts)

A growing number of passive acoustic monitoring systems have resulted in a wealth of annotation information, or metadata, for recordings. These metadata are semi-structured. Some parameters are essentially mandatory (e.g., time of detection and what was detected) while others are highly dependent upon the question that a researcher is asking. Tethys is a metadata system for spatial-temporal acoustic data that provides structure where it is appropriate and flexibility where it is needed. Networked metadata are stored in an extended markup language (XML) database, and served to workstations over a network. The ability to export summary data to OBIS-SEAMAP is in development. The second purpose of Tethys is to serve as a scientific workbench. Interfaces are provided to networked databases, permitting the import of data from a wide variety of sources, such as lunar illumination or sea ice coverage. Interfaces currently exist for MATLAB, JAVA, and PYTHON. Writing data driven queries using a single interface enables quick data gathering from multivariate sources to address hypotheses. Examples showing the results of analysis of acoustic data from acoustic deployment from 26 sites across the Northern Pacific will be shown.

**3:15–3:40 Break**

## *Invited Papers*

**3:40**

**4pAB7. Long-term bioacoustic monitoring for gauging animal populations: An approach involving flight calls of night migrating songbirds.** William Evans (Old Bird Inc., 605 W. State St., Ithaca, NY 14850, admin@oldbird.org)

At times during the course of a year, the airspace over most locations in North America contains flight calls of night migrating songbirds. The calls typically have an audio frequency between 2 and 10 kHz and are 0.03-0.4 s in duration. Documenting such calls in a consistent manner enables temporal and quantitative calling patterns to be determined. Theoretically, such acoustic data gathered over time could be used as an index to population change as well as for documenting shifting migration routes. This presentation will discuss the development of a multi-sensor system designed to synchronously sample nocturnal flight calls of migrating songbirds across eastern North America. We will review the decisions involved with the system design that minimize non-target aspects of the soundscape and that help standardize monitoring over time. We will then illustrate the monitoring power of this application with flight call data from ten fall migrations at one monitoring station and two fall migrations from a transect of ten monitoring stations across eastern North America.

4:00

**4pAB8. Acoustic monitoring of breeding amphibians at Yosemite National Park and Point Reyes National Seashore.** Patrick Kleeman, Gary Fellers (US Geological Survey, 1 Bear Valley Rd., Point Reyes, CA 94956, pkleeman@usgs.gov), and Brian Halstead (US Geological Survey, Dixon, CA)

The calling behavior of frogs and toads at breeding sites lends itself to acoustic monitoring of these amphibian populations. We are using Automated Recording Devices (ARD) at two National Park units in California, Yosemite National Park and Point Reyes National Seashore, to monitor the breeding efforts of amphibians by recording their calls. We are monitoring both common species (Pacific chorus frogs, *Pseudacris regilla*, occurs in both parks) and imperiled species (Yosemite toad, *Anaxyrus canorus*, Yosemite; California red-legged frog, *Rana draytonii*, Point Reyes) to investigate whether breeding phenology will shift with changing climatic conditions. The use of ARD is also providing a more complete picture of the diel calling patterns of these species, and how some species are partitioning their acoustical environment by frequency and time in order to breed successfully while surrounded by noisy neighbors. Information gathered through acoustic monitoring is very valuable for conserving rare amphibians and ensuring that common species remain common.

## *Contributed Papers*

4:20

**4pAB9. Determining the contribution of cetacean noise to the marine soundscape of Australia's Northwest Shelf.** Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au)

The continental shelf off Western Australia is 500 km wide and has an annual mean sea surface temperature of 28 degrees Celsius. Biodiversity is rich and includes at least 45 cetacean species (whales and dolphins). A catalog of sounds produced by these animals was established based on a literature review and recordings obtained by the Centre for Marine Science and Technology at Curtin University. An automatic detector for these sounds was developed and includes the following steps: (1) Fourier transformation of the recorded time series, (2) spectrogram normalization, (3) computation of information entropy of the spectrogram, (4) investigation of entropy distribution and thresholding, and (5) removal of detections with fewer than a predefined number of spectrogram pixels. Detector performance was assessed by comparison to manual detections. The total energy of the biological sounds detected was computed to determine the contribution of cetaceans to the underwater noise budget at several locations and times of year. [Work supported by Chevron Australia.]

4:35

**4pAB10. Spatiotemporal variability in coral reef soundscapes in St. John, U.S. Virgin Islands.** Maxwell B. Kaplan, T. Aran Mooney (Biology Dept., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS50, Woods Hole, MA 02543, mkaplan@whoi.edu), and Jim Partan (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Passive acoustic measurements of coral reef "soundscapes" can be an effective way of tracking biological activity and may help assess community-level biotic diversity. While a reef soundscape may vary both temporally and spatially, this variability is often not well understood. To investigate this, we deployed multiple digital acoustic recorders (DMONs) for both short- (24-h) and long-term (4 months) investigations at three patch reefs that varied in coral cover (low, intermediate, and high levels) in the U.S. Virgin Islands National Park (sample rate: 120 kHz). The short-term investigation consisted of four continuously recording instruments spaced at ~20 m intervals. Long-term measures included two recorders per reef on a duty cycle of 2.5 min/2 h. Fish and coral diversity, ambient light intensity, temperature, and salinity were also measured. Results indicate diel patterns in snapping shrimp signals (dominant energy between 2.5 and 20 kHz) with peaks at dusk and dawn. Sound pressure level (SPL) of the snapping shrimp band varied spatially within and among reefs, with higher maximum SPL at reefs with low and intermediate coral cover. However, within-reef SPL variability was lowest at the site with high coral cover. Temporal patterns in snapping shrimp acoustic activity were correlated within and among all three reefs.

4:50

**4pAB11. The ambient acoustic environments at two locations in Laguna San Ignacio, Baja, Mexico.** Kerri Seger (Scripps Inst. of Oceanogr., Univ. of California San Diego, 4090 Rosenda Ct, Unit 199, San Diego, CA 92122, kseger@ucsd.edu), Aaron M. Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Steven Swartz (Laguna San Ignacio Ecosystem Sci. Program, Darnestown, MD), and Jorge Urban (Laboratorio de Mamíferos Marinos, Universidad Autonoma de Baja California Sur, La Paz, Mexico)

Each winter gray whales (*Eschrichtius robustus*) breed and calve in Laguna San Ignacio, with the lagoon's northern section more heavily used by mothers rearing calves. The southern section of the lagoon is open to milling and ecotourism traffic, while the northern section is restricted to vessel transits only. Ambient acoustic data from autonomous underwater recorders have been collected between 2008 and 2013 at Punta Piedra (southern section) and Camp Kuyima (northern section). Multiple sources of acoustic sound exist in the lagoon, including tidal flows, fish chorusing, gray whale vocalizations, snapping shrimp, daily land/sea breezes, and panga activity. Here the cumulative distributions of rms sound pressure levels from all deployments during daytime and nighttime are presented for several frequency bands that represent contributions from the varying source mechanisms. Since concurrent data from both restricted (northern) and unrestricted (southern) sections exist, comparing sound level distributions between sites can provide insight into the relative contributions of various mechanisms to the overall ambient noise environment. These data have established a baseline for monitoring trends and changes in acoustic environments of the lagoon in anticipation of future tourist development. [Work sponsored by LSIESP and Ocean Foundation.]

# Session 4pAO

## Acoustical Oceanography and Animal Bioacoustics: Properties, Trends, and Utilization of Ocean Noise II

Jennifer L. Miksis-Olds, Cochair
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Zoi-Heleni Michalopoulou, Cochair
*Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102*

**Chair's Introduction—1:55**

### *Invited Paper*

**2:00**

**4pAO1. Soundscapes from hydrophone stations in the comprehensive nuclear-test-ban treaty organisation's international monitoring system hydroacoustic network.** Mark K. Prior and David Brown (IDC/SA, Comprehensive Nuclear-Test-Ban Treaty Organisation, PO Box 1200, Vienna 1400, Austria, mark.prior@ctbto.org)

The Comprehensive Nuclear-Test-Ban Treaty Organisation operates a global network of sensors that includes cabled sound-channel hydrophones in the Atlantic, Pacific and Indian Oceans. Hydrophones are deployed in groups of three, known as triads, so that the arrival times and azimuths of signals can be obtained. Data are recorded at frequencies up to 100 Hz with continuous acquisition and data relay via satellite connection to CTBTO's International Data Centre. Signals from distant earthquakes, underwater explosions, marine mammals and ice-breaking are routinely detected and an extensive archive has been built up over the last decade. To understand sensor detection performance, high-level summaries of noise properties are required to establish the "acoustic context' for each station. These "soundscapes" allow the identification of source types that dominate in specific frequency bands. Examples signals are illustrated and information regarding the sources of persistent signals is extracted.

### *Contributed Papers*

**2:20**

**4pAO2. The marine soundscape of the Fram Strait.** Hanne Sagen, Hans Kristian Tengesdal, Mohamed Babiker (Nansen Environ. and Remote Sensing Ctr. (NERSC), Bergen, Norway), and Dag Tollefsen (Norwegian Defence Res. Establishment (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no)

The marine soundscape of the Fram Strait has been subject to investigations since the mid 1980's. Increasing interest in Arctic operations has initiated a recent series of acoustic experiments that includes synoptic ambient noise measurements in the Marginal Ice Zone conducted over the years 2010–2012. This presentation will give an overview of these experiments, then focus on results from measurements made with sonobuoys under varying ice and environmental conditions in the MIZ. Noise spectra (20 Hz—2 kHz) are presented, discussed, and compared with historical data from 1985 to 1987. Spectra are categorized by environmental parameters including wind force and direction as derived from numerical models, ice concentration derived from satellite images, ocean wave properties from a coupled ice-ocean prediction model, and sound propagation conditions inferred from the ice-ocean model. The contributions to this soundscape that will be quantified and discussed include open-ocean wind-generated noise, ice floe collision, marine mammals, and seismic exploration activity.

**2:35**

**4pAO3. Using an autonomous underwater vehicle to track the changing arctic ambient noise field in real time.** Stephanie Fried and Henrik Schmidt (Massachusetts Inst. of Technol., 77 Massachusetts Ave, Rm 5-204, Cambridge, MA 02139, eowyn@mit.edu)

The ambient noise field, particularly the directionality of the noise, can provide a wealth of information about the local environment. Changes in the ambient noise field often reflect changes in the physical environment. Accurate calculation of the noise field, though, can be a challenge. Because of their maneuverability autonomous underwater vehicles (AUVs) provide novel capabilities not only for measuring and analyzing the local noise field, but also for continuous tracking of changes to the noise field and thus the environment. Of interest is the measurement and analysis of the ambient noise in arctic environments. By integrating models for arctic ambient noise into an AUV simulation, this paper analyzes the use of AUVs in real-time autonomous tracking of the three-dimensional changing arctic ambient noise field.

### 2:50

**4pAO4. Interdecadal trends in ocean ambient sound at seven sites in the northern Pacific Ocean basin.** Rex K. Andrew (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98040, rex@apl.washington.edu), Bruce M. Howe (Ocean & Res. Eng., Univ. of Hawaii-Manoa, Honolulu, HI), and James A. Mercer (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

A long-term observation program begun in approximately 1994 has amassed time-series of ambient sound short-time spectra from omni-directional hydrophones deployed on the ocean floor at seven locations in the northern Pacific Ocean. Each time-series consists of spectra estimated every 5 min over a useful band of 10–500 Hz, thereby encompassing the vocalizations of baleen whales, the anthropogenic contribution of ship traffic noise, and wind/wave noise due to sea surface processes. Simple linear trend lines show that traffic noise in the northern and northwestern reaches of the Pacific has increased by as much as 3 to 4 dB during this program. In the eastern North Pacific, however, the ambient sound shows a decrease of about 3 dB. The number of ships in the world merchant fleet increased by approximately 25% over this period. This change is insufficient to explain the increases in noise levels along northern and northwestern regions, and provides no explanation for the decreases observed in the north-east. The traffic noise field is evidently dependent on more complex temporal and geographical patterns of shipping traffic. [Work supported by ONR.]

## *Invited Paper*

### 3:05

**4pAO5. Quantifying ocean noise and its spatiotemporal variability on Australia's Northwest Shelf.** Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au), Alexander Gavrilov, and Robert McCauley (Ctr. for Marine Sci. & Technol., Curtin Univ., Perth, WA, Australia)

The Northwest Shelf is an extensive oil and gas region off Western Australia. The Centre for Marine Science and Technology at Curtin University has recorded underwater noise in this region for 14 years on behalf of industry and government. Under the Collaborative Environmental Research Initiative (CERI), this data is being shared and synthesized to quantify the marine soundscape and to describe spatiotemporal variability. Automatic software analysis tools were developed to process the data. Power spectrum density percentiles were computed for all sites on a monthly basis, and compared. Distinct spectral features were identified. Factors contributing to the observed spatiotemporal variability ranged from long-term offshore oil and gas installations to fish choruses. [Work supported by Chevron Australia.]

## *Contributed Papers*

### 3:25

**4pAO6. Wind dependence of shallow water ambient noise in a biologically rich temperate coastal area.** Delphine Mathias (GIPSA-Lab, 11 rue des Mathématiques, Saint Martin d'Hères 38402, France, delphine.mathias@gmail.com), Cedric Gervaise, and Lucia Di Iorio (GIPSA-Lab, Chair Chorus Grenoble INP Foundation, Saint Martin d'Hères, France)

The Iroise Marine Natural Park, created in 2007, is the first French natural marine park. This archipelago located in Western Brittany is a shallow water area that comprises 11 islands and hosts a rich variety of marine life, including seaweed fields, benthic organisms, endangered seals, and cetaceans. Three underwater autonomous recorders were moored at 10-m depth and sampled at 32 kHz from June 2011 to November 2011. Here we report on the dependency of shallow water ambient noise level on wind speed in a biologically rich environment. First we extract the ambient noise level in presence of transient sounds produced by benthic organisms by removing instantaneous sound pressure levels higher than a threshold computed using the kurtosis of the raw 10 sec time series. We then show that the ambient noise level allows to extract environmental information such as wind speed and biological rhythms, and that both are explaining 90% of its variance. Dependence of ambient noise and ocean noise level to wind speed at several frequencies are compared to reference work by Wenz (1962) and previous shallow water studies. Finally, we discuss how data assimilation coupling measured ambient noise and environmental parameters can help monitor marine ecosystems.

### 3:40

**4pAO7. Correlating acoustical with physical and biological oceanography.** Iain Parnum (Ctr. for Marine Sci. & Technol., Curtin Univ., Perth, WA, Australia), Christine Erbe (Ctr. for Marine Sci. & Technol., Curtin Univ., Kent St., Bentley, WA 6102, Australia, c.erbe@curtin.edu.au), and Arti Verma (Ctr. for Marine Sci. & Technol., Curtin Univ., Perth, WA, Australia)

The Centre for Marine Science & Technology at Curtin University built and maintains the underwater acoustic recorders of Australia's Integrated Marine Observing System (IMOS; http://IMOS.org.au). Recordings have been obtained at four locations (off Western Australia, Victoria, and New South Wales) since 2011. IMOS includes a multitude of oceanographic and remote sensors, contributed by various institutions, which are also responsible for data management. Data are shared and publicly available encouraging collaboration and syntheses. This study has compiled time series of weather data, tides, current data (from Acoustic Doppler Current Profilers, ADCP), and wind (from radar measurements), and established correlations with underwater noise is a series of one-third octave bands between 10 Hz and 3 kHz from the Perth Canyon. Our results further demonstrate that ocean noise in certain frequency bands can be used to estimate aspects of physical and biological oceanography.

### 3:55

**4pAO8. Undersea noise characterization using vector sensors and adaptive particle filters.** Dennis Lindwall (Phys., Univ. of New Orleans, 2000 Lakeshore Dr., New Orleans, LA 70148, lindwall@bellsouth.net), Don DelBalzo (Marine Information Resource Corp., Ellicot City, MD), Dimitrios Charalampidis (Elec. Eng., Univ. of New Orleans, New Orleans, LA), Jim Leclere, E. J. Yoerger, and George Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

Acoustic noise in the ocean is spatially complex and dynamic. It results from a composite of moving discrete and distributed sources with narrow and broadband signatures from near and distant locations over complex propagation paths. Whether the goal of noise characterization is to improve understanding of the sources and environmental aspects of noise or to find and describe weak signals, it can be better achieved with vector acoustic data and vector-based analysis rather than with a pure scalar-pressure approach. We will show analytically and with realistic acoustic noise simulations that the vector characterization of noise parameters such as directional distribution, frequency content, and time variation is more accurate and can be done with simpler instrumentation than what is commonly done with pressure data. We employ adaptive particle filters to model and estimate the temporal aspects of the noise fields. The filter's state and observation vectors consist of signal directions and magnitudes, respectively. As a result, the noise fields are directly associated with the filter's process and observation noise.

4:10

**4pAO9. Ocean wave seismic and acoustic noise detected with distributed fiber optic sensor array.** Kent K. Hathaway (Coastal & Hydraulics Lab., US Army Engineer Res. & Development Ctr., 1261 DC Rd., Kitty Hawk, NC 27949, Kent.K.Hathaway@usace.army.mil), Richard D. Costley, Eric Smith, Troy Milburn, and Jennifer R. Picucci (GeoTech. & Structures Lab., U.S. Army Engineer Res. & Development Ctr., Vicksburg, MS)

The U.S. Army Engineer Research & Development Center installed a Fiber Optic Sensor System (FOSS) at the Field Research Facility (FRF) in Duck, NC as a test-bed for evaluating FOSS performance in measuring ambient acoustic and seismic noise in a coastal environment. The sensor system consists of a buried fiber optic cable with a length of approximately 3 km, which wends its way through the sand dunes and along the beach. An optical interrogator contains a pulsed laser which injects light into an optical fiber within the cable and receives Rayleigh backscattered signals from it. The system, which is able to detect and locate seismic activity along the entire length with a 10 m resolution, is being used to study ocean wave generated noise. The FRF also maintains and operates a real-time cross-shore directional wave array, consisting of five bottom-mounted acoustic wave gauges installed at 2 to 11 m depths. The main focus of the work presented here compares FOSS data with directional wave measurements made with the cross-shore array under a variety of conditions. Data collected with other sensor systems (e.g., vertical long-period seismometer, infrasound microphone, anemometers, rain gauges, high resolution video) were also used in this comparison.

THURSDAY AFTERNOON, 5 DECEMBER 2013 GOLDEN GATE 2/3, 1:00 P.M. TO 6:00 P.M.

## Session 4pBA

## Biomedical Acoustics: Recent Advances in Therapeutic Ultrasound II

Tatiana D. Khokhlova, Cochair
*Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105*

Oleg Sapozhnikov, Cochair
*Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation*

### *Contributed Papers*

1:00

**4pBA1. Laparoscopic high intensity focused ultrasound for the treatment of soft tissue.** Narendra T. Sanghvi and Adam Morris (R & D, SonaCare Medical, 4000 Pendleton Way, Indianapolis, IN 46226, narensanghvi@sonacaremedical.com)

There is a growing demand to perform focal surgery, particularly for prostate and kidney tumors. RF and Cryo ablation are used for treatment of tumors; however, there are reported complications mainly skipping of cancer cells and bleeding. It has been shown that HIFU can provide acoustic hemostasis to overcome such complications. However, HIFU must compete with these modalities with improved treatment efficiency and the size of the applicator must fit in a 12 mm trocar/port used during the laparoscopic surgery. We modified the Sonatherm-600i applicator and transducer that generates a split beam HIFU at 4 MHz with a center element for imaging operating at 6.5 MHz. The transducer is integrated in a robotic probe that renders bi-plane images for tissue localization and ablation. Based on selected tissue ablation volume, treatment trajectory is generated by the computer to provide optimum thermal dose to result in complete coagulative necrosis. The treatment is conducted using continuous HIFU with an interlaced imaging for tissue change monitoring. The ablation efficiency of this device is @ 1 cc/min that is better than Cryo and RF ablation. Results will be presented from recent animal studies.

1:15

**4pBA2. Model-based feasibility assessment and evaluation of prostate hyperthermia with a commercial MRI-guided endorectal high intensity focused ultrasound ablation array.** Vasant A. Salgaonkar (Radiation Oncology, Univ. of California San Francisco, 396 Ano Nuevo Ave., Apt. 106, Sunnyvale, CA 94085, salgaonkarv@radonc.ucsf.edu), Viola Rieke, Eugene Ozhinsky (Radiology and Biomedical Imaging, Univ. of California San Francisco, San Francisco, CA), Punit Prakash (Elec. and Comput. Eng., Kansas State Univ., Manhattan, KS), Juan Plata (Radiology, Stanford Univ., Stanford, CA), John Kurhanewicz (Radiology and Biomedical Imaging, Univ. of California San Francisco, San Francisco, CA), I-C (Joe) Hsu, and Chris J. Diederich (Radiation Oncology, Univ. of California San Francisco, San Francisco, CA)

Numerical simulations were conducted to devise methods for targeted and protracted hyperthermia (40–46 °C, 30–60 min) to the prostate with a commercial MR-guided endorectal ultrasound phased array (2.3 MHz, ExAblate, InSightec). The intention is to fast-track clinical implementation of this FDA approved ablation system for delivering targeted hyperthermia in conjunction with radiation or chemotherapy. Conformable hyperthermia to focal tumors in posterior and hemi-gland prostate was simulated through 3D patient-specific biothermal models and beamformed acoustic patterns that incorporated the specific constraints imposed on the ExAblate array:

irregular element spacing, switching speeds, operating power and short pulse duration. Simulations indicated that diverging and iso-phase sonications could treat ($T > 41\,^\circ$C, $_{max} < 46\,^\circ$C) 13–23 $cm^3$ with ~1.1 $W/cm^2$, multi-focused patterns could treat 4.0 $cm^3$ with 3.4 $W/cm^2$, and curvilinear patterns could treat 6.5 $cm^3$ with 0.8 $W/cm^2$ while avoiding rectum, urethra, pubic bone, *etc*. Custom beamforming identified through simulations was implemented on the ExAblate system and sonications were performed in tissue mimicking phantom material. ExAblate delivered long duration sonications (0.86 $W/cm^2$) with these customized beamforming patterns and generated diffuse hyperthermia ($\Delta T = 4$–$8\,^\circ$C) in phantom, monitored with real-time multi-slice MR temperature imaging (3T). [NIH R01CA122276, Focused Ultrasound Foundation.]

**1:30**

**4pBA3. Ultrasound-mediated remote actuation of implantable devices for localized drug delivery.** Parag V. Chitnis (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, pchitnis@riversideresearch.org), Olga Ordeig, and Samuel K. Sia (Dept. of Biomedical Eng., Columbia Univ., New York, NY)

Direct local delivery of therapeutics can significantly improve long-term outcome and quality of life for cancer patients. We present a drug-delivery approach that employs focused ultrasound (FUS) for remotely actuating drug-loaded, biocompatible implants consisting of porous NiPAAm hydrogels encapsulated in a PDMS disk. The hydrogels were 1 mm thick and 6 mm in diameter. The NiPAAm formulation was designed to contract to 30% of original size when heated to 45°C. The capsule was loaded with 20-kDa TRITC-Dextran and placed in a custom-designed PDMS chamber containing de-ionized water. A thermocouple was embedded in the NiPAAm gel to monitor local temperature. TRTIC-dextran released to the surrounding media was quantified by absorbance at 540/580 nm. Maintaining the capsules at 37°C for two days using a hot plate did not trigger release. A 1.5-MHz FUS transducer operating at low intensities (<500 $W/cm^2$) elevated the gel temperature to 45°C in $32.6 \pm 19$ s ($N = 10$). Thermocouple was employed as a feedback to modulate (on/off) FUS in real-time to maintain gels at 45°C for 10~min, which resulted in a release of $10.6 \pm 0.3$~$\mu$g of dextran. Capsules were then implanted in eight mice; four mice were subjected to FUS. FUS actuated release *in vivo* as evidenced by fluorescence imaging.

**1:45**

**4pBA4. High intensity focused ultrasound laparoscopic instrument for partial nephrectomy.** Stuart Mitchell, Yak-Nam Wang (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, sbmitch@apl.washington.edu), Jonathan Harper, Ryan Hsi (Urology, Univ. of Washington Medical Ctr., Seattle, WA), and Lawrence Crum (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Partial nephrectomy (PN) is the gold standard for small clinically localized renal masses because of equal oncologic outcomes and greater preservation of renal function compared with radical nephrectomy (RN). However, it is a complex operation due to the challenges of cutting into a well-vascularized organ and the need for reconstruction of the remaining kidney following excision. PN is associated with higher blood loss, risk of transfusion, and longer operative time compared to RN. High intensity focused ultrasound (HIFU) affords the ability to ablate tissue and perform hemostasis, thus potentially mitigating some of the challenges associated with PN. The purpose of this paper is to introduce a new HIFU clamp as an adjunctive tool for PN. A HIFU device was created to conform to the shape of a commonly used laparoscopic instrument. Characterization studies were conducted using *ex vivo* tissue. Histology was performed to evaluate thermal damage. *Ex vivo* studies indicated that complete ablation planes could be achieved at temperatures sufficient for thermal tissue necrosis. Gross parenchymal changes were observed with clear demarcation between treated and untreated regions. Histological evaluations revealed that there were no viable cells in ablated regions. [This work was funded by NIH (EB013365).]

**2:00**

**4pBA5. Pulsed focused ultrasound treatment of muscle mitigates paralysis-induced bone loss in the adjacent bone: A study in a mouse model.** Sandra L. Poliachik (Dept. of Radiology, Seattle Children's Hospital, Seattle, WA), Tatiana D. Khokhlova, Yak-Nam Wang, Julianna C. Simon (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, tanyak@apl.washington.edu), Ted S. Gross (Dept. of Orthopaedics, Univ. of Washington, Seattle, WA), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA)

Bone loss can occur following bed rest, space flight, spinal cord injury, or age-related hormonal changes. The treatment methods for this condition include pharmaceutical interventions and exercise, neither of which is particularly effective. Other technologies include low intensity pulsed ultrasound targeted to the bone, used previously to enhance fracture healing, and whole body vibration. This study attempted to mitigate paralysis-induced bone loss indirectly, by applying pulsed focused ultrasound (pFUS) to the midbelly of a paralyzed muscle. We employed a mouse model of disuse that utilizes onabotulinumtoxin A, which induces rapid bone loss in 5 days. The pFUS treatments were performed daily for four consecutive days following paralysis. A spherically focused 2-MHz transducer produced 5-microsecond pulses at pulse repetition frequency mimicking motor neuron firing rates during walking (80 Hz) or standing (20 Hz). Two different power levels were used corresponding to peak positive focal pressures of 30 and 18 MPa. The trabecular bone changes were characterized using micro computed tomography. Our results indicated that application of pFUS at pulse repetition frequency of 20 Hz and lower amplitude setting successfully mitigated paralysis-induced bone loss. The targeted muscle tissue did not display any sign of injury. [Work supported by CDMRP SCIRP (SC090510).]

**2:15**

**4pBA6. Optimization of parameters for therapeutic applications of high intensity myocardial contrast echocardiography.** Douglas Miller, Chunyan Dou (Radiology, Univ. of Michigan, 3240A Medical Sci. I, 1301 Catherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu), Gabe E. Owens (Pediatric Cardiology, Univ. of Michigan, Ann Arbor, MI), and Oliver D. Kripfgans (Radiology, Univ. of Michigan, Ann Arbor, MI)

High intensity myocardial contrast echocardiography (HI-MCE) can lethally injure cardiomyocytes leaving scattered microlesions. This cavitational bioeffect may be of value for graded tissue-reduction therapy for conditions such as hypertrophic cardiomyopathy. Anesthetized rats in a heated water bath were treated with 1.5 MHz focused ultrasound, which was guided by an 8 MHz diagnostic imaging probe. Eight-pulse bursts were triggered intermittently over 5 min at approximately end systole during contrast microbubble infusion. The relative efficacy between 2 MPa (~173 $W/cm^2$ $I_{PA}$) or 4 MPa (~892 $W/cm^2$) pulses, 1:4 or 1:8 trigger intervals, and 5 or 10 cycle pulses was explored in 6 groups. ECG premature complexes (PCs) induced by the triggered pulse bursts were counted, and microlesions assessed in Evans blue-stained cardiomyocyte scores (SCSs). The increase from 2 to 4 MPa produced significant increases in PCs and SCSs. In addition, the higher pressure eliminated the decline in the rate of PC induction over time, which was seen at 2 MPa and likely hindered therapeutic efficacy. Neither increased trigger intervals nor pulse durations yielded significant increases in therapeutic effects. High concentrations of microlesions were readily produced, which suggest that HI-MCE can be refined into a clinically robust method for therapeutic myocardial reduction.

**2:30**

**4pBA7. Delivery of different-size molecules by ultrasound-induced blood-brain barrier opening and its correlation with acoustic emission.** Hong Chen, Anushree Srivastava, Tao Sun, Oluyemi Olumolade, and Elisa E. Konofagou (Dept. of Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, hc2666@columbia.edu)

Focused ultrasound (FUS) in combination with microbubbles (MBs) has been successfully used in the delivery of therapeutic agents of various sizes through the blood-brain barrier (BBB) in preclinical studies. However, the dependence of delivery efficiency on the drug molecular size calls for further exploration. Fluorescence-labeled dextrans of molecular weights of 3, 70, 150, and 2000 kDa were used as model therapeutic compounds. Dextrans were mixed with MBs and injected intravenously to mice immediately after the onset of FUS sonication. The acoustic emission from ultrasound-activated MBs was acquired passively using a 10 MHz transducer and quantified for stable, inertial, and total cavitation doses. The drug delivery efficiency was quantified by the relative delivery amount and volume estimated based on fluorescent images of the brains. It was found that dextran of 3 kDa can be delivered trans-BBB at a pressure level below the inertial cavitation threshold; however, dextrans of 70, 150, and 2000 kDa were delivered only when the pressure was above the inertial cavitation threshold. At the same pressure level, the amount and volume of dextrans delivered decreased as the dextran size increased. A linear correlation of total cavitation dose and the fluorescence enhancement was found for each size dextran.

**2:45**

**4pBA8. Synergy between high intensity focused ultrasound and ethanol injection in ablation of thyroid cancer cells.** Hakm Murad (Biomedical Eng., TuLn. Univ., 108 Cottonwood Dr., Gretna, LA 70056, hmurad@tulane.edu), Nguyen H. Hoang, Sithira H. Ratnayaka (Biomedical Eng., TuLn. Univ., New Orleans, LA), Koji Tsumagari, Emad Kandil (Surgery, TuLn. Univ. School of Medicine, New Orleans, LA), and Damir Khismatullin (Biomedical Eng., TuLn. Univ., New Orleans, LA)

We have investigated the combination of high intensity focused ultrasound (HIFU) and ethanol injection for ablation treatment of anaplastic thyroid cancer, a highly aggressive form of thyroid cancer characterized by >80% mortality within months. The suspension of FB1 anaplastic thyroid cancer cells (100 $\mu$l, 2.7 million cells/ml) were placed in a 0.2 ml thin-wall PCR tube and then exposed to HIFU alone, ethanol alone, or ethanol + HIFU. The focused ultrasound signal was generated by a 1.1 MHz transducer with acoustic power ranged from 4.1 W to 12.0 W. Ethanol was diluted in the FB1 cell growth medium to the concentration of 2%, 4%, or 10% (v/v) and applied to the cells before HIFU exposure. The viability of the cells was measured by flow cytometry and trypan blue exclusion 2, 24, and 72 h post-treatment. The exposure of FB1 cells to HIFU alone greatly reduced the number of viable cells immediately after treatment; however, their proliferation rate remained high. On the other hand, both the viability and proliferation rate significantly decreased in the cells treated with both ethanol and HIFU. In conclusion, percutaneous ethanol injection (PEI) and HIFU have a synergistic effect on anaplastic thyroid cancer ablation.

**3:00**

**4pBA9. High intensity focused ultrasound ablation of ethanol-treated liver tissues and cancer cells.** Nguyen H. Hoang (Biomedical Eng., TulaneUniv., 5243 Beaucaire St., New Orleans, LA 70129, nhoang@tulane.edu), Hakm Y. Murad (Biomedical Eng., TulaneUniv., Gretna, LA), Sithira H. Ratnayaka, Chong Chen, and Damir B. Khismatullin (Biomedical Eng., TulaneUniv., New Orleans, LA)

We have investigated the combined effect of HIFU and ethanol injection on the temperature rise and cavitation in porcine liver tissues and on the viability and proliferation rate of HepG2 liver cancer cells. Tissues were injected with 95% ethanol before being subjected to the HIFU beam generated by a 1.1 MHz transducer with acoustic power ranged from 1.17 W to 20.52 W. Cavitation events and the temperature in and around the focal zone were measured by a passive cavitation detector and type K thermocouples, respectively. In the cell study, 100 $\mu$l of HepG2 cell suspension (2.7 million cells/ml) was placed in a 0.2 ml thin-wall PCR tube. Ethanol 2% or 4% in the cell growth medium was added to the cell suspension, and the cells were then exposed to HIFU for 30 s. The data of these experiments show that the pre-treatment of tissues with ethanol reduces the threshold power for inertial cavitation and increases the temperature rise. Both the viability and proliferation rate were significantly decreased in cells treated with ethanol and HIFU, as compared to individual treatments. The results of our study indicate that ethanol injection and HIFU have a synergistic effect on liver cancer ablation.

**3:15–3:30 Break**

**3:30**

**4pBA10. High intensity focused ultrasound for *Enterococcus faecalis* biofilm.** Siew-Wan Ohl (Fluid Dynam., Inst. of High Performance Computing, 1 Fusionopolis Way, #16-16 Connexis North, Singapore 138632, Singapore, ohlsw@ihpc.a-star.edu.sg), Kulsum Iqbal (Discipline of Prosthodontics, Operative Dentistry and Endodontics, National Univ. of Singapore, Singapore, Singapore), Boo Cheong Khoo (Dept. of Mech. Eng., National Univ. of Singapore, Singapore, Singapore), Jennifer Neo (Discipline of Prosthodontics, Operative Dentistry and Endodontics, National Univ. of Singapore, Singapore, Singapore), and Amr Sherif Fawzy (Discipline of Oral Sci., National Univ. of Singapore, Singapore, Singapore)

High intensity focused ultrasound (HIFU) is used to removal *Enterococcus faecalis* (*E. faecalis*) in planktonic suspension and dental biofilm. The bacteria *E. faecalis* is commonly found in secondary dental infection after root canal treatment. Sealed petri dish with *E. faecalis* planktonic suspension is placed at the focal region of the bowl-shaped HIFU transducer of 250 kHz in a water bath. It is subjected to sonification of 30 to 120 s. It is found that the HIFU successfully lysed and removed the bacteria from counting its colony forming units (CFU), performing scanning electron microscopy (SEM) and confocal microscopy. Also, *E. faecalis* biofilms in human teeth are subjected to the same HIFU treatment. Similar analysis is performed with SEM and confocal microscopy. It is found that after 60 s of sonification, most of the biofilm is either removed or lysed. In conclusion, this study highlights the potential of using HIFU as non-destructive dental root canal disinfection treatment.

**3:45**

**4pBA11. Accurate quantification and delivery of thermal dose to cells in culture.** Elly Martin, Adam Shaw, Nilofar Faruqui, and Michael Shaw (Acoust. and Ionizing Radiation Div., National Physical Lab., Hampton Rd., Teddington, Middlesex TW11 0LW, United Kingdom, adam.shaw@npl.co.uk)

HIFU treatments involve raising the temperature of target tissue above 60°C in short (~2 s) bursts. At higher temperatures, shorter times are required to induce a given deleterious effect: the Sapareto-Dewey thermal dose equation is often used to relate the time to produce a biological effect at one temperature to the time to produce equivalent effects at another. A heating chamber was developed to deliver controlled thermal doses to cells in culture under continual observation by differential interference contrast microscopy. The system comprised of a cell culture well and cover slip coated with a transparent electrode inserted into a microscope stage with electrical contacts. Thermal doses were delivered by applying programmed current-time profiles and using a PID controller to rapidly raise and maintain the temperature of the chamber above 37°C while monitoring with fine wire thermocouples. Initially, HeLa cells in monolayer culture were imaged before, during, and after heating. Visible changes in cell shape and adhesion began shortly after raising the temperature by 8°C and progressed during a heating period of 20 min, continuing for more than 12 h after the cells were returned to 37 °C. No such changes were observed in control cells. Results will be presented exploring the validity of the S-D relationship for shorter, higher temperature exposures.

**4:00**

**4pBA12. Thermal lesion imaging using Lorentz force: Proofs of concept.** Pol Grasland-Mongrain, Stefan Catheline (LabTAU, INSERM U1032, 151 Cours Albert Thomas, Lyon 69424, France, jean-yves.chapelon@inserm.fr), Jean-Martial Mari (Imperial College, London, France), Rémi Souchon, Ali Zorgani, Alexandre Petit, Florian Cartellier, Cyril Lafon, and Jean-Yves Chapelon (LabTAU, INSERM U1032, Lyon, France)

The Lorentz force can be used by different means to image thermal lesions in biological tissue. In the first method presented here, so-called magneto-acoustical electrical tomography, a tissue sample is held in a magnetic field and is subsequently exposed to a focused ultrasound beam. The displacement within the magnetic field caused by this ultrasound beam results in an electrical current due to the Lorentz force. In this way, the change in electrical conductivity due to the presence of thermal lesions can then be observed. Conversely, when an electrical current is applied to tissue placed in a magnetic field, a shear wave is induced by the Lorentz force.

Elastography images can be reconstructed from this shear wave, revealing thermal lesions by the change in elastic modulus. The first method was tested on *ex-vivo* chicken breast sample with a 500 kHz transducer and a 300 mT magnetic field. The second method was tested on gelatin phantom with a 100 mA current and 300 mT magnetic field. Images and results will be presented for both methods. These techniques could be used for the monitoring of thermal lesion formation in high intensity focused ultrasound treatment.

4:15

**4pBA13. Prediction of the reversibility of the ultrasound-induced blood-brain barrier opening using passive cavitation detection with magnetic resonance imaging validation.** Tao Sun, Gesthimani Samiotaki, and Elisa E. Konofagou (Dept. of Biomedical Eng., Columbia Univ., 630 West 168th St., Physicians & Surgeons 19-418, New York, NY 10032, ts2765@columbia.edu)

Various molecules have been shown to cross the blood-brain barrier (BBB) upon exposure to focused ultrasound combined with microbubbles and exhibit therapeutic effects. Real-time monitoring, thereof, remains one of the key elements before clinical translation of this technique. The dependence of acoustic emissions on the closing timelines of the BBB opening volume and its permeability was investigated under different pressures (0.30, 0.45, and 0.60 MPa) and microbubble sizes (diameters: 1–2, 4–5, or 6–8 $\mu$m). A 10-MHz passive cavitation detector was used to acquire cavitation signals during sonication at the mouse right hippocampus ($n = 45$). Contrast-enhanced dynamic and T1-weighted MR scans were performed immediately after sonication and up to 6 days thereafter. Contrast-enhanced volumes and diffusion rates of the contrast agent were quantified as indicators for the BBB opening. It was found that the stable cavitation dose increased with the number of days required for closing while it reached a plateau after day 4. However, the inertial cavitation dose exhibited an exponential increase with the duration of the opening. A linear correlation between the total cavitation dose and BBB opening days was found. Moreover, the volume and permeability indicator $K_{\text{trans}}$ were found to be both pressure- and bubble size-dependent. The dependence on the bubble-diameter and pressure allows us to predict and control the safety profile of this technique.

4:30

**4pBA14. Rapid aberration correction for transcranial magnetic resonance-guided focused ultrasound surgery using a hybrid simulation and magnetic resonance-acoustic radiation force imaging method.** Urvi Vyas and Kim Butts Pauly (Radiology, Stanford Univ., 1420 Guerrero St., San Francisco, CA 94110, urvivyas@stanford.edu)

Transcranial magnetic resonance-guided focused ultrasound surgery is a technique for causing tissue necrosis in the brain though the intact skull. Skull spatial and acoustic heterogeneities cause changes in the location, shape, and intensity of the focus. Current techniques use computed tomography (CT) imaging or MR-acoustic radiation force images (MR-ARFI) to correct these aberrations. CT-based techniques approximate acoustic parameters from Hounsfield units but suffer from co-registration concerns. MR-ARFI-based techniques use MR images as feedback to manipulate transducer phases, but require many image acquisitions (~4000) for one correction [Herbert, IEEE-TUFFC **56**(11)2388–2399]. We demonstrate here a hybrid technique that uses one MR-ARFI image to improve the focal intensity. The hybrid simulation-MR-ARFI technique used an optimization routine to iteratively modify the simulation aberrations to minimize the difference between simulated and experimental radiation force patterns. Experiments were conducted by applying skull-based aberrations to a 1024-element, 550 kHz phased-array transducer. The experimental MR-ARFI image of the aberrated focus was used with the simulation pattern from the hybrid angular spectrum [Vyas, IEEE-TUFFC **59**(6)1093–1100] beam propagation technique to estimate aberrations. The experiment was repeated three times. The hybrid simulation-MR-ARFI technique resulted in an average increase in focal MR-ARFI phase of 44%, and recovered 83% of the ideal MR-ARFI phase.

4:45

**4pBA15. A non-axisymmetric, elongated pressure distribution in the lithotripter focal plane enhances stone comminution *in vitro* during simulated respiratory motion.** Jaclyn M. Lautz, Georgy Sankin, Joseph Kleinhenz, and Pei Zhong (Mech. Eng. & Mater. Sci., Duke Univ., Sci. Dr., Durham, NC 27708, jaclyn.lautz@duke.edu)

A challenge in clinical shock wave lithotripsy (SWL) is stone translation due to a patient's respiratory motion, in a direction perpendicular to shock-wave propagation, which may negatively affect stone comminution while increasing the risk of tissue injury. We have developed a method using external masks and a modified lens geometry to transform the axisymmetric pressure distribution in the focal plane of an electromagnetic lithotripter into a non-axisymmetric elliptical distribution. At equivalent acoustic pulse energy (46 mJ), the peak pressure was reduced from 44 MPa to 38 MPa while the $-6$ dB focal width was increased from 7.4 mm for the original to 11.7 mm (major axis) and 7.9 mm (minor axis) of the modified field. *In vitro* stone comminution was performed in a tube holder ($d = 14$ mm) using a translation pattern with 12 breaths per minute and 15 mm in excursion distance. Stone comminution after 1000 shocks are $71.2 \pm 4.4\%$ and $65.2 \pm 8.3\%$ ($p < 0.05$) along the major- and minor-axis of the modified field, respectively, compared to $62.6 \pm 7.2\%$ for the original axisymmetric field. These results suggest that an elongated pressure field aligned along the direction of stone motion may enhance stone comminution in SWL. [Work supported by the NIH and the NSF GRFP.]

5:00

**4pBA16. Shockwave tensile phase transmission depends on the gas concentration of the coupling medium.** Spencer T. Frank (Mech. Eng., Univ. of California Berkeley, 1849 Cedar St., Apt. B, Berkeley, CA 94703, spencerfrank@berkeley.edu), Jaclyn Lautz, Georgy N. Sankin, Pei Zhong (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC), and Andrew Szeri (Mech. Eng., Univ. of California Berkeley, Berkeley, CA)

Previous research shows that a shockwave's tensile phase can be strongly attenuated as a function of gas concentration in the coupling medium. Here, we seek to elucidate the relationship between tensile attenuation and gas concentration via pressure measurements at the focus and highspeed imaging. By performing *in vitro* experiments with water of varying gas concentrations (2.05 mg/L, 4.30 mg/L, and 6.50 mg/L), the negative impulsive pressure is correlated to the density of the bubble cloud that occurs in the beampath. It is found that for gas contents below 4 mg/L the bubble cloud remains sparse and the shockwave's tensile phase is successfully transmitted with no loss in impulsive pressure. For gas contents 4 mg/L and above the bubble cloud becomes highly dense and prevents transmission with up to a 75% loss in impulsive pressure. Corresponding stone comminution experiments show that the treatment efficiency sharply decreases with increasing gas concentration. These results underlie the importance of degassing the water used in the coupling medium before treatment.

5:15

**4pBA17. Fragmentation of kidney stones *in vitro* by focused ultrasound bursts without shock waves.** Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, 1013 NE 40th St., Seattle, WA 98105, amax38@u.washington.edu), Bryan W. Cunitz, Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Ryan S. Hsi, Mathew D. Sorensen, Jonathan D. Harper (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Shock wave lithotripsy (SWL) is the most common procedure for treatment of kidney stones. SWL noninvasively delivers high-energy focused shocks to fracture stones into passable fragments. We have recently observed that lower-amplitude, sinusoidal bursts of ultrasound can generate similar fracture of stones. This work investigated the characteristics of stone fragmentation for natural (uric acid, struvite, calcium oxalate, and cystine) and artificial stones treated by ultrasound bursts. Stones were fixed in position in a degassed water tank and exposed to 10-cycle bursts from a

200-kHz transducer with a pressure amplitude of $p \leq 6.5$ MPa, delivered at a rate of 40–200 Hz. Exposures caused progressive fractures in the stone surface leading to fragments up to 3 mm. Treatment of artificial stones at different frequencies exhibited an inverse relationship between the resulting fragment sizes and ultrasound frequency. All artificial and natural types of stones tested could be fragmented, but the comminution rate varied significantly with stone composition over a range of 12–630 mg/min. These data suggest that stones can be controllably fragmented by sinusoidal ultrasound bursts, which may offer an alternative treatment strategy to SWL. [Work supported by NIH 2T32DK007779-11A1, R01 EB007643, P01 DK043881, R01 DK092197, NSBRI through NASA NCC 9-58.]

**5:30**

**4pBA18. Kidney stone fracture by surface waves generated with focused ultrasound tone bursts.** Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, olegs@apl.washington.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Wayne Kreider, Bryan W. Cunitz, and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Previous studies have provided insight into the physical mechanisms of stone fracture in shock wave lithotripsy. Broadly focused shocks efficiently generate shear waves in the stone leading to internal tensile stresses, which in concert with cavitation at the stone surface, cause cracks to form and propagate. Here, we propose a separate mechanism by which stones may fragment from sinusoidal ultrasound bursts without shocks. A numerical elastic wave model was used to simulate propagation of tone bursts through a cylindrical stone at a frequency between 0.15 and 2 MHz. Results suggest that bursts undergo mode conversion into surface waves on the stone that continually create significant stresses well after the exposure is terminated. Experimental exposures of artificial cylindrical stones to focused burst waves *in vitro* produced periodic fractures along the stone surface. The fracture spacing and resulting fragment sizes corresponded well with the spacing of stresses caused by surface waves in simulation at different frequencies. These results indicate surface waves may be an important factor in fragmentation of stones by focused tone bursts and suggest that the resulting stone fragment sizes may be controlled by ultrasound frequency. [Work supported by NIH 2T32DK007779-11A1, R01 EB007643, P01 DK043881, R01 DK092197, NSBRI through NASA NCC 9-58.]

**5:45**

**4pBA19. Histotripsy beyond the "intrinsic" cavitation threshold using very short ultrasound pulses: "Microtripsy".** Kuang-Wei Lin, Yohan Kim (Biomedical Eng., Univ. of Michigan, 2200 Bonisteel Blvd., Gerstacker, Rm. 1107, Ann Arbor, MI 48109, kwlin@umich.edu), Adam D. Maxwell (Urology, Univ. of Washington, School of Medicine, Seattle, WA), Tzu-Yin Wang (Radiology, Stanford Univ., Stanford, CA), Timothy L. Hall, Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), Brian Fowlkes (Radiology, Univ. of Michigan, Ann Arbor, MI), and Charles A. Cain (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI)

Conventional histotripsy uses pulses with $\geq$3 cycles wherein the bubble cloud formation relies on the pressure-release scattering of the positive shock fronts from sparsely distributed cavitation bubbles. In a recent work, the peak negative pressure (P(-)) threshold for the generation of dense bubble clouds directly by a negative half cycle were measured, and this threshold has been called the "intrinsic threshold." In this work, characteristics of lesions generated with this intrinsic threshold mechanism were investigated using RBC phantoms and excised canine tissues. A 32-element, PZT-8, 500 kHz therapy transducer was used to generate short (<2 cycles) histotripsy pulses at PRF = 1Hz and P(-) = 24.5–80.7 MPa. The results showed that the spatial extent of the histotripsy-induced lesions increased as the applied P(-) increased, and the lesion sizes corresponded well to the estimates of the focal regions above the intrinsic threshold. The sizes for the smallest reproducible lesions averaged $0.9 \times 1.7$mm (lateral $\times$ axial), significantly smaller than $-6$ dB beamwidth of the transducer ($1.8 \times 4.0$ mm). These results suggest that predictable, well-confined and microscopic lesions can be precisely generated using the intrinsic threshold mechanism. Since the supra-threshold portion of the negative half cycle can be precisely controlled, lesions considerably less than a wavelength are easily produced ("microtripsy").

THURSDAY AFTERNOON, 5 DECEMBER 2013 — MASON, 1:30 P.M. TO 4:00 P.M.

# Session 4pEA

## Engineering Acoustics: Beam Control of Microphone and Transducer Arrays

Michael Zarnetski, Chair
*NUWC, Newport, RI 02841*

### *Contributed Papers*

**1:30**

**4pEA1. Microphone array exploratory study.** Marc Messier (Univ. of Miami, 1527 Albenga Ave., University Village Bldg. 4 Rm. 401A, Coral Gables, FL 33146, m.messier@umiami.edu)

In the field of acoustic signal processing, one of the most popular areas of research is that of microphone arrays. As a small research project in this field, a microphone array system with adjustable polar response will be developed. It is worth noting that this submission is more a research proposal than an abstract. This research will take part as a means of combining coursework and research for courses in engineering acoustics and real-time digital signal processing at the University of Miami. Before any arrays are physically constructed or any code written on a DSP, simulations will be performed in MATLAB to determine optimum array configurations and to analyze various DSP algorithms for adjusting polar responses. Then, using a Texas Instruments TMS320C6713 DSP on a DSK audio target board, and writing code in C, Assembly, and MATLAB, physical tests will be conducted to analyze performance of design. A MEMS microphone model will likely be used for its small size and low cost. From there, results will be analyzed and further research in the area proposed.

**1:45**

**4pEA2. Prediction of the spatial response of a microphone capsule using scattering and lumped-element simulations.** Douglas Rollow (Technol. and Innovation, Sennheiser Electronic Corp, 550 15th St., Ste. 37, San Francisco, CA 94103, tad.rollow@sennheiser.com), Vladimir Gorelik, Meike Wulkau (Technol. and Innovation, Sennheiser Electronic GmbH & Co. KG, Wedemark, Germany), and Sebastian Chafe (Technol. and Innovation, Sennheiser Electronic Corp., San Francisco, CA)

When a microphone capsule is placed in an environment with surrounding structures, the electrical response of the capsule will be dictated by the

capsule design and by acoustic scattering of the structures surrounding it. Lumped element models of the transducer are typically used in predicting the electrical response to the field, but when the transducer is operating at a frequency where the spatial variation of the field is significant, the simplifying assumptions used in these models no longer hold. In this work, a finite element-based scattering simulation provides blocked-port field quantities to drive a lumped element circuit model, predicting the electrical output as a function of the frequency and incident angle of an incoming plane wave. The scattering simulation allows for the inclusion of mechanical supporting structures and protective screens, and their influence on the field at the rear port of a gradient transducer. Simulated results are shown for the simplified capsule and compared to measurements of a real capsule in free field conditions as well as with surrounding structures.

**2:00**

**4pEA3. Microphone array with computer vision based directivity.** Marc Messier and Jordan Reimers (Univ. of Miami, 1527 Albenga Ave., University Village Bldg. 4 Rm. 401A, Coral Gables, FL 33146, m.messier@umiami.edu)

In the field of acoustic signal processing, one of the most popular areas of research is that of microphone arrays. As a small research project in this field, a microphone array system with adjustable polar response will be developed. To make research more innovative and multi-disciplinary, the polar response of the array will be controlled by a facial tracking system implemented with computer vision techniques. This research will take part as a means of combining coursework and research for courses in engineering acoustics, computer vision, and real-time digital signal processing at the University of Miami. Before any physical testing, simulations will be performed in MATLAB to determine optimum array configurations and to analyze various facial tracking and dsp algorithms. Then, using a Texas Instruments TMS320C6713 DSP on a DSK audio target board, and writing code in C, ASSEMBLY, and MATLAB, physical tests will be conducted to analyze performance of design. A MEMS microphone model will likely be used for its small size and low cost. A Microsoft Kinect and compatible desktop computer will be used for the computer vision interface. From there, results will be analyzed and further research in the area proposed.

**2:15**

**4pEA4. A simple adaptive cardioid direction finding algorithm.** Gary W. Elko (mh Acoustics LLC, 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens Meyer (mh Acoustics LLC, Fairfax, Vermont)

A simple direction-finding algorithm using three or four omnidirectional microphones is described. The algorithm is based on the minimization of the output power of a generally steerable cardioid microphone. It is shown that the algorithm can be reduced to running three independent single-tap LMS filters for the general 3D case and two independent single-tap LMS adaptive filters for the 2D (null constrained to lie in a plane). Results will also be shown for a 32-element spherical microphone array for sources corrupted with microphone self-noise and reverberation.

**2:30**

**4pEA5. *In situ* evaluation of surround sound system performance.** Eric M. Benjamin (Surround Res., 1229 Springwood Way, Pacifica, CA 94044, ebenj@pacbell.net), Aaron J. Heller (Artificial Intelligence Ctr., SRI Int., Menlo Park, CA), and Fernando Lopez-Lezcano (Ctr. for Comput. Res. in Music and Acoust., Stanford Univ., Palo Alto, CA)

Surround sound systems are produced with the intention of reproducing the spatial aspects of sound, such as localization and envelopment. As part of his work on Ambisonics, Gerzon developed two metrics, the velocity and energy localization vectors, which are intended to predict the localization performance of a system. These are used during the design process to optimize the decoder that supplies signals to the loudspeaker array. At best, subjective listening tests are conducted on the finished system, but no objective assessments of the spatial qualities are made to verify that the realized performance correlates the predictions. In the present work, binaural recordings were made of a 3-D 24-loudspeaker installation at Stanford's Bing Studio. Test signals were used to acquire the binaural impulse response of each loudspeaker in the array and of Ambisonic reproduction using the loudspeaker array. The measurements were repeated at several locations within the hall. Subsequent analysis calculated the ITDs and ILDs for all cases. Initial results from the analysis of the ITDs and ILDs for the center listening position show ITDs, which correspond very closely to what is expected in natural hearing, and ILDs, which are similar to natural hearing.

**2:45**

**4pEA6. Network modeling of multiple-port transducers across multiple modes of vibration.** Michael L. Kuntzman, Nishshanka N. Hewa-Kasakarage, Donghwan Kim, and Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, 10100 Burnet Rd., Bldg. 160, MER 1.108, Austin, TX 78752, mlkuntzman@gmail.com)

A network modeling procedure is presented that is capable of modeling transducers across a broad frequency regime with multiple coupling ports. The model is based on modal superposition, and a separate network is crafted for each vibration mode of the device. Modal velocity, rather than a particular physical velocity on the vibrating transducer, is chosen as the flow variable in each network. Multiple ports are modeled with the use of multiple transformers in series. A procedure for performing system identification to complete the network parameters is also presented, which can be performed experimentally, analytically, or through use of a finite element model in the design stage. Application of the procedure to a multiple port piezoelectric microphone is presented.

**3:00**

**4pEA7. Environment mapping and localization with an uncontrolled swarm of ultrasound sensor motes.** Erik Duisterwinkel (INCAS3, Dr. Nassaulaan 9, Assen 9401 HJ, Netherlands, erikduisterwinkel@incas3.eu), Libertario Demi, Gijs Dubbelman (Dept. of Elec. Eng., Eindhoven Univ. of Technol., Eindhoven, Netherlands), Elena Talnishnikh, Heinrich J. Wörtche (INCAS3, Assen, Netherlands), and Jan W. Bergmans (Dept. of Elec. Eng., Eindhoven Univ. of Technol., Eindhoven, Netherlands)

A method is presented in which a (large) swarm of sensor motes perform simple ultrasonic ranging measurements. The method allows to localize the motes within the swarm, and at the same time, map the environment which the swarm has traversed. The motes float passively uncontrolled through the environment and do not need any other sensor information or external reference other than a start and end point. Once the motes are retrieved, the stored data can be converted into the motes relative positions and a map describing the geometry of the environment. This method provides the possibility to map inaccessible or unknown environments where electro-magnetic signals, such as GPS or radio, cannot be used and where placing beacon points is very hard. An example is underground piping systems transporting liquids. Size and energy constraints together with the occurrence of reverberations pose challenges in the way the motes perform their measurements and collect their data. A minimalistic approach in the use of ultrasound is pursued, using an orthogonal frequency division multiplexing technique for the identification of motes. Simulations and scaled air-coupled 45–65 kHz experimental measurements have been performed and show feasibility of the concept.

**3:15**

**4pEA8. Simulations about non-Doppler continuous wave usage of ultrasonic transducers.** Emre İkizler (Informatics and Information Security Res. Ctr. (BILGEM), The Sci. and Technolog. Res. Council of Turkey (TUBITAK), TUBITAK Yerleskesi BILGEM Binasi, Gebze 41400, Turkey, emre.ikizler@tubitak.gov.tr) and Hulya Sahinturk (Dept. of Mathematical Eng., Yildiz Tech. Univ., Istanbul, Turkey)

In this work, computer simulations are executed in order to detect the presence of an object which passes through the area illuminated by ultrasonic transmitter. In simulations, object which has a constant speed can be detected by only observing received signal level on ultrasonic receiver, not by using Doppler Effect or Fast Fourier Transform. Additionally, alterations on speed and distance of the object causes sensible alterations on received signal level on ultrasonic receiver. According to simulation results, a simple way for detecting speed of objects is able to be introduced by using more simple signal processing techniques compared to Doppler Effect or Fast Fourier Transform related techniques.

**3:30**

**4pEA9. Development of a multi-resonance transducer for highly directional underwater communication.** Yonghwan Hwang, Yub Je (Mech. Eng., Postech, PIRO416, Postech, Hyo-ja dong, Nam gu, Po hang KS010, South Korea, serenius@postech.ac.kr), Jaeil Lee, Jonghyeon Lee (Ocean System Eng., Jeju Univ., Jeju, South Korea), Wonho Kim, Heesun Seo (ADD, Jin hae, South Korea), and Wonkyu Moon (Mech. Eng., Postech, Po hang, South Korea)

The parametric array is a nonlinear conversion process that generates a narrow beam of low-frequency sound using small aperture. It can be applied to underwater communication between two nodes with known locations, since the highly directional sound beam may provide such benefits as privacy, no interference due to the multi-path. The difference frequency wave (DFW) from the parametric array shows small side lobes and extraordinary directivity. The shortcoming of the DFW generated by the parametric array may be its low sound pressure level relative to that of the directly generated sound beams. In this study, we designed and fabricated a multi-resonance transducer as a parametric array source and evaluated its feasibility as a transmitter. For that purpose, we determined the proper design parameters for midrange communication. We selected 10 kHz as the communication frequency and then determined the primary frequencies as 100 and 110 kHz. We composed the source transducer using the two kinds of unit transducers. The fabricated transducer array and the developed operating techniques enabled us to successfully transmit letters, words, and drawings inside the water tank. By testing the characteristics, we confirmed that the developed operating scheme and transducer can be used for underwater communication. [Work supported by ADD (UD130007DD).]

**3:45**

**4pEA10. A basic study on frequency characteristics compensation of sound at ear drum when using hearing aids.** Hitoshi Iseda (Dept. of Mech. and Aerosp. Eng., Tokyo Inst. of Technol., Chofugaoka 3-66-1,801, Chofu, Tokyo 182-0021, Japan, iseda.h.aa@m.titech.ac.jp) and Masaaki Okuma (Dept. of Mech. and Aerosp. Eng., Tokyo Inst. of Technol., Kawagoe, Japan)

Wearing the acoustic equipment such as hearing aids (HAs) and earphones changes the frequency characteristics of sound at the ear drum because the equipment influences acoustic condition of outer ear canal as a partition wall in the canal. This influence should be compensated in order not to degrade the quality of auditory perception. To achieve this, the identification of acoustic property in the canal is required for both cases of with and without HAs. The final goal of this research is to develop a universal method for the frequency characteristics compensation of sound at the ear drum when wearing HAs and incorporating the algorithm of the method into such equipment. In this paper, as the first phase of the research, the authors present the results of an experimental identification technique using simple and large scale models of canal and HAs based on the theory of the transfer matrix method. The result of a compensation experiment using digital filter processing is also presented.

THURSDAY AFTERNOON, 5 DECEMBER 2013 CONTINENTAL 1, 2:00 P.M. TO 4:30 P.M.

## Session 4pMUa

## Musical Acoustics and Structural Acoustics and Vibration: Acoustics of Percussion Instruments II

Thomas D. Rossing, Chair
*Stanford Univ., Stanford, CA 94022*

### *Invited Papers*

**2:00**

**4pMUa1. Modeling orchestral crotales as thin plates.** Thomas R. Moore, Daniel W. Zietlow, and Donald C. Griffin (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

Orchestral crotales are designed in such a way that the overtones become less harmonic as the fundamental pitch increases. Deutsch, *et al.* used Kirchhoff-Love theory to show that by eliminating the central mass and choosing the correct ratio of inner to outer radius the overtones can be harmonically related (J. Acoust. Soc. Am. **116**, 2427 (2004)). However, when a crotale was constructed using this design, the overtones were not harmonically related. We show that this lack of agreement between theory and experiment is due to the fact that shear motion of the inner boundary, which is neglected in classical thin-plate theory and thought to be unimportant, can significantly affect the resonance frequencies of plates even when they are extremely thin.

**2:20**

**4pMUa2. Acoustics of Western and Eastern bells, old and new.** Thomas D. Rossing (CCRMA, Stanford Univ., 26464 Taaffe Rd., Los Altos Hills, CA 94022, rossing@ccrma.stanford.edu)

The modes of vibration and sound radiation from tuned church bells, carillon bells, handbells, ancient Chinese two-tone bells, and temple bells will be compared. Most bells have a circular cross section, but many ancient Chinese bells do not, and thus they have two different strike notes, depending upon where they are struck. The musical interval between these two strike notes is often near a minor third or a major third.

## *Contributed Papers*

**2:40**

**4pMUa3. Evolution of the Hang percussion instrument and associated performance practices.** David Wessel (Music CNMAT, Univ. of California Berkeley, 1750 Arch St., Berkeley, CA 94709, davidwessel@me.com) and Thomas Rossing (Thomas Rossing, CCRMA, Stanford, CA)

It is now over 10 years since the widespread adoption of the Hang percussion instrument. It has evolved considerably since its invention in 2000 by Felix Rohner and Sabina Scharer in Bern, Switzerland. We present acoustical analyses of variations of the instrument in conjunction with an evolving performance technique. The talk will include demonstrations of performance practices and their acoustical consequences.

**3:00**

**4pMUa4. Characterization of the mechanical properties of the steelpan.** April Bryan, Marc Gobin, Akill Griffith, Dillon Frederick (Dept. of Mech. and Manufacturing Eng., The Univ. of the West Indies, St. Augustine, Trinidad and Tobago, aprilbr@gmail.com), Brian Copeland (Elec. and Comput. Eng., The Univ. of the West Indies, St Augustine, Trinidad and Tobago), and Clement Imbert (Mech. and Manufacturing Eng., The Univ. of the West Indies, St Augustine, Trinidad and Tobago)

The steelpan is a struck idiophone whose playing surface is constructed by forming the top of a fifty-five gallon steel oil drum into a sunken, nearly hemispherical surface and then raising smaller shells on the hemisphere to form notes. The completed instrument resembles an inverted turtle shell and is played by striking the notes with sticks. Although it is understood that variations in note geometry and material properties are mainly responsible for the characteristic sounds generated when the notes are struck, few studies have investigated these relationships. Previous research efforts have explored the metallurgical properties and the characteristic vibrations of the notes. Less emphasis has been placed on the relationship between the mechanical properties of the steelpan and its acoustic behavior. In this research, the variation in the mechanical properties across the playing surface of a tenor steelpan is characterized. Of the instruments in the steelpan family, this instrument has the greatest deformation and the most notes. More specifically, the variation in the residual stress, strength, Young's Modulus, and Poisson's ratio are determined and compared among octave sets. This characterization is important for the development of models that relate the mechanical properties to the acoustical behavior of the steelpan.

## *Invited Papers*

**3:20**

**4pMUa5. Rhythmic techniques and psychoacoustic effects of the percussion music of Steve Reich.** Garry M. Kvistad (Woodstock Percussion, Inc., 167 DuBois Rd., Shokan, NY 12481, garry@chimes.com)

Beginning in the 1960s, composer Steve Reich began to experiment with rhythmic devises. "Phase shifting" is where unison melodies become canons as one voice speeds up slightly. This was done first with tape loops and later with live performers. In 1967, he applied this technique to "Piano Phase" in which two pianists on two pianos start a pattern in unison. One player slowly speeds up, creating canons which lock in and out of rhythmic "unisons" eventually ending back in melodic unison. This process continues two more times with different patterns. This rhythmic technique was used in several other early works of Reich such as his 1970 composition, "Drumming." NEXUS member Garry Kvistad has built a multi-element set of percussion instruments tuned in just intonation to perform Piano Phase. The first section is played on a large wooden bar instrument similar to an African Amadinda. The second section is played on a set of thick aluminum tubes. The last section is played on an aerophone activated by slapping the ends of closed tubes. In this presentation, Mr. Kvistad will demonstrate this difficult performance technique using the newly built instruments.

**3:40**

**4pMUa6. A first look into the caxirola—Official music instrument of the Soccer World Cup 2014.** Talita Pozzer and Stephan Paul (UFSM, Undergraduate Program in Acoust. Eng., DECC-CT-UFSM, Av. Roraima 1000, Camobi, Santa Maria, RS 97105-900, Brazil, talita.pozzer@eac.ufsm.br)

For the 2014 Soccer World Cup Brazilian Musician Carlinhos Brown created the caxirola as the official music instrument, adapting an old African instrument—the caxixi. Both caxixi and caxirola generate sound by hard particles impacting on the walls of a closed basket. While the basket of the caxixi is made of natural materials the basket of the caxirola is made of environmentally friendly polymer. Remembering the acoustical impact of the vuvuzela in 2010 as quite negative it was decided to study the acoustics of the caxirola. First, the way people will use the caxirola in terms of arm's excursion and velocity of shaking was studied. It was found that the caxirola was used moving it longitudinally and perpendicularly to its main axis, both in vertical direction. Spectra of the sound emitted by the caxirola and caxixi moving perpendicularly are quite similar with slightly more low frequency energy. However, when subjected to longitudinal movements, spectra are different. For the caxirola, the particle impact sound on the walls is the same for all impacted walls but for the caxixi, where lateral walls are of different material than the bottom, the spectra shows more energy for impacts at bottom.

### *Contributed Papers*

**4:00**

**4pMUa7. Javanese gong wave signals.** Matias H. Budhiantho and Gunawan Dewantoro (Electronics and Comput. Eng., Satya Wacana Christian Univ., Jln. Diponegoro 52-60, Salatiga, Jawa Tengah 50711, Indonesia, mhwb@Gmail.com)

In Central Java, the Gong is one of eminent gamelan instrument, an ensemble of predominantly struck instruments that has deep philosophical meaning for Javanese. However, there lack of studies concerning on this particular instrument as a bridging means between scientific description and human artistic perception. This study aims to investigate the spectral and temporal properties as well as particularly look into the typical wave-like sound of the Gong. Acoustic measurements were conducted and analyzed using ARTA. Both frequency and time domain analyses were explored to better understand the nature of the Gong wave signals. The fundamental frequency which decays much more slowly than the other harmonic started with lower increasing frequency. The wave-like sound of the Gong maybe due to signal behavior the resemblance beat phenomenon between early and later development of the fundamental sustaining frequency.

**4:15**

**4pMUa8. Percussion, via transducers.** Alex Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The sounds within a music recording are necessarily mediated by the microphones and loudspeakers associated with their recording and playback. The effect of microphones—type and placement—invites a unique view of musical acoustics. The necessity of loudspeaker playback motivates a range of signal processing strategies, using equalization, compression, reverberation, distortion, and more to further reshape the sound. This paper reviews the sonic influence of contemporary audio engineering craft on the sound of percussion instruments as realized in music recordings.

THURSDAY AFTERNOON, 5 DECEMBER 2013 CONTINENTAL 1, 5:00 P.M. TO 6:00 P.M.

## Session 4pMUb

## Musical Acoustics: Percussion Concert

Thomas D. Rossing, Chair
*Stanford Univ., Stanford, CA 94022*

A mini-concert will be held following session 4pMUa featuring Gary Kvistad, David Wessel, Punita Singh, Rohan Krishna Murthy and others.

THURSDAY AFTERNOON, 5 DECEMBER 2013 CONTINENTAL 9, 1:00 P.M. TO 5:25 P.M.

## Session 4pNS

## Noise and Structural Acoustics and Vibration: Active Control of Sound and Vibration

Scott D. Sommerfeldt, Cochair
*Dept. of Physics, Brigham Young Univ., N181 ESC, Provo, UT 84602*

Kenneth Cunefare, Cochair
*Georgia Technol., Mech. Eng., Atlanta, GA 30332-0405*

### *Invited Papers*

**1:00**

**4pNS1. Active noise control: Eight decades of research and applications.** Kenneth Cunefare (School of Mech. Eng., The Georgia Inst. of Technol., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

On January 27, 1933, Paul Lueg submitted his patent application for an active noise control system. In the intervening 80 years, the concept has progressed from simple single-input single-output feedback control systems through to complex MIMO approaches implementing a seemingly endless variety of control algorithms. Academic publications appear to continue at approximately a constant pace over the past decade, with much attention paid to algorithm refinement and development of applications. Commercial successes of active noise control include ANC headset in the aviation and consumer markets, as well as systems installed in passenger vehicle (some of which also incorporating active vibration control systems). This paper will provide a brief historical retrospective on the development of active noise control, and survey the current state of academic research as well as existing and proposed production applications (consumer, aviation, defense, etc.).

**1:20**

**4pNS2. Strategies for improving speech intelligibility and warning signal detection in communication headsets/hearing protectors.** Anthony J. Brammer, Eric R. Bernstein, and Gongqiang Yu (Ergonomics Technol. Ctr., UConn Health Ctr., 263 Farmington Ave., Farmington, CT 06030-2017, brammer@uchc.edu)

Strategies for improving speech understanding and warning signal detection when wearing communication headsets/hearing protectors (HPDs) in environmental noise must accommodate sounds from different sources at different times. A subband signal processing approach would appear desirable, with a delayless structure essential for active noise reduction (ANR). The requirements for communication channel and ANR controllers differ, owing to the different bandwidths required for speech and warning signals, and for ANR. Subbands for optimizing speech signal-to-noise ratios are commonly fractional-octave bandwidth, while computational efficiency favors linear subbands for ANR. Increasing the number of subbands reduces computational cost for the latter but the advantage is less apparent for communication signal control. Possibilities exist for harmonizing subband filter structures by constructing models of speech intelligibility using computationally efficient bandwidths. In contrast, algorithms for detecting warning sounds tend to be governed more by audibility than bandwidth considerations. The issues will be discussed using a simulation of a circumaural HPD that can replicate word scores obtained by a subject when subject-specific transfer functions are employed. Where available, results from physical devices and subjects will be included, as well as the consequences of differences in individual auditory abilities. [Work supported by NIOSH grant R01 OH008669.]

**1:40**

**4pNS3. A perceptually motivated active noise cancellation system for hearing impaired listeners: An overview.** Buye Xu, Jinjun Xiao, and Tao Zhang (Signal Processing Res., Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com)

Nowadays most hearing impaired (HI) patients are fitted with either open-fitting or vented-fitting hearing aids (HAs) to reduce the occlusion effect. In an environment where strong low-frequency ambient noise is present, significant noise energy can directly leak into the ear canal bypassing noise reduction algorithms in the HAs and may reduce speech intelligibility and listening comfort for HA users. One way to mitigate such an issue without occluding the ear canal is to implement an active noise cancellation (ANC) system inside the ear canal. Traditional ANC systems are designed to minimize the total sound pressure level in the ear canal. However, this may not necessarily lead to an optimal solution from the perceptual perspective (e.g., loudness may be reduced instead of being minimized as a result). In this paper, a perceptually motivated feedback ANC system is presented: a spectral shaping filter is applied to the residual error signal to minimize the loudness for HI listeners. In addition, implications of the practical constraints introduced by the HAs are discussed based on acoustic simulations and experimental results.

**2:00**

**4pNS4. A better frequency domain adaptive algorithm for active noise control in a short duct.** Jing Lu and Xiaojun Qiu (Inst. of Acoust., Nanjing Univ., Hankou Rd. 22th, Nanjing 210093, China, lujing@nju.edu.cn)

For the feedforward active noise control in a short duct, the noncausality of the whole system is often inevitable, since the reference sensor needs to be placed very close to the control source, and the acoustic transmission delay between them is often surpassed by the inherent AD/DA and anti-aliasing filters latency in the controller. The commonly used normalized frequency domain LMS algorithm possesses the benefit of fast convergence speed, but suffers from deteriorated steady-state performance when used in non-causal circumstances. In this paper, an efficient modification of the normalized frequency domain LMS algorithm is proposed, which can significantly improve wide-band noise reduction level in non-causal circumstances. Both simulations and experiments demonstrate the superiority of the proposed algorithm.

**2:20**

**4pNS5. Local and global active noise control using a parametric array loudspeaker.** Nobuo Tanaka (Tokyo Metropolitan Univ., 6-6 Asahigaoka, Hino-shi, Tokyo 191-0065, Japan, ntanaka@sd.tmu.ac.jp)

This paper deals with local as well as global active noise control (ANC) using a parametric array loudspeaker (PAL) possessing intriguing properties: sharp directivity, low sound pressure decay by distance, capability of steering directivity, etc. After briefing some properties of a PAL necessary for ANC, this paper presents pinpoint control using a PAL for suppressing the sound pressure at a designated location, hence local control. It is shown that unlike conventional ANC in which a voice coil loudspeaker is used, the pinpoint control may achieve the sound pressure suppression without causing spillover. Using the same sound control source, PAL, this paper then refers to global control, termed trivial control in the art, enabling one to generate a global zone of quiet. The trivial control strategy falls into a category of acoustic power control based on a trivial condition requiring the collocation of a primary source and a control source. The trivial condition formerly avoided because of literally trivial may be implemented due to the property of a PAL, thereby enhancing the control effect, theoretically infinity. Two kinds of control strategy are then demonstrated with a view to fulfilling the trivial condition for global control.

**2:40**

**4pNS6. Virtual mechanical impedance approach for the active structural acoustic control of panels.** Alain Berry, Marc Michau, Philippe Micheau (Mech. Eng., Université de Sherbrooke, 5907 Laurent, Sherbrooke, QC J1N 3Z2, Canada, alain.berry@usherbrooke.ca), and Philippe Herzog (Laboratoire de Mécanique et d'Acoustique, Marseille, France)

This work investigates harmonic Active Structural Acoustic Control of flexural panels using structural, collocated and dual actuator-sensor pairs. Two types of transducer technologies are envisioned: (1) thin piezoelectric actuators and sensors; (2) electrodynamic inertial actuators and transverse velocity sensors. The control strategy is to locally impose a complex, virtual mechanical impedance to the structure via a linear relation between the actuator input and sensor output of each pair, at each frequency of interest. This virtual impedance is optimized to minimize the sound radiation of the structure at the corresponding frequency. The approach is implemented as a two-step process: (1) the optimal virtual impedance matrix is derived from identification of the primary sound and transfer functions between the control actuators, structural sensors and far-field acoustic sensors; (2) the optimal virtual impedance matrix is imposed using a real-time, iterative controller. Numerical and experimental results are discussed to highlight the underlying physical interpretation of the virtual impedances for sound radiation or sound transmission control. The implication of the different actuator and sensor technologies in terms of sensing the global vibration and acoustic response is also discussed.

**3:00–3:15 Break**

**3:15**

**4pNS7. Active structural acoustic control using a sum of weighted spatial gradients control metric.** William R. Johnson, Daniel R. Hendricks, Monty J. Anderson, Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT 84602, will.johnson@byu.edu), and Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Active structural acoustic control (ASAC) is an active noise control technique, which provides global control by targeting and minimizing the structural vibrations which contribute to radiated sound power. The majority of research in ASAC has focused on validating various proposed concepts on flat rectangular plates, an important but not comprehensive class of structures. To extend the body of knowledge, ASAC has been investigated on finite ribbed plates under a variety of boundary conditions. Simulated results have shown that two different approaches, minimizing the volume velocity and minimizing the weighted sum of spatial gradients (WSSG) provide comparable average attenuation of radiated sound power on ribbed plates. With regards to sensing, minimizing WSSG has several advantages over minimizing volume velocity. In particular, WSSG has been formulated to be easier to measure than volume velocity, without requiring a priori information about the structure or its modes. WSSG has also been shown to be relatively uniform spatially and relatively insensitive to boundary conditions, while also providing improved control over volume velocity at structural modes higher than the first mode. These results suggest that more practical, complex vibrating structures can be effectively controlled for the reduction of radiated sound power using the WSSG approach.

**3:35**

**4pNS8. Convergence analysis of filtered-x least mean squares algorithm for active control of repetitive impact noise.** Guohua Sun, Tao Feng, Mingfeng Li, and Teik C. Lim (College of Eng. and Appl. Sci., Univ. of Cincinnati, 801 ERC, P.O. Box 210018, 2901 Woodside Dr., Cincinnati, OH 45221, teik.lim@uc.edu)

The prevalent adaptive active noise control (ANC) algorithm, namely, the filtered-x least mean squares (FXLMS), exhibits a critical challenge for treating transient impact noise. This is because the FXLMS algorithm requires certain adaptation time to converge satisfactorily. However, the FXLMS algorithm may have its learning capacity when the transient noise shows certain repeatability. In this paper, a distinctive theoretical analysis of the convergence behavior of ANC system with the standard FXLMS algorithm is conducted for repetitive impulse-induced transient noise control. To simplify the derivation, the secondary path is assumed to be a pure delay model. Through this analysis, a step size bound condition is derived, and an optimal step size that leads to the fastest convergence is determined. To validate the analysis, extensive numerical simulations are performed considering various pure delay secondary path models. Calculations are in very good agreement with the theoretical analysis. Finally, a more general secondary path is considered to further demonstrate the effectiveness of the FXLMS algorithm for repetitive impulse noise control. The results indicate that ANC system with the FXLMS algorithm can be a very promising technique for repetitive transient noise typically seen in industrial facilities such as punching machines.

## *Contributed Papers*

**3:55**

**4pNS9. Active control of sound transmission through soft-cored sandwich panels using volume velocity cancellation.** Kiran C. Sahu and Prof. Jukka Tuhkuri (Dept. of Appl. Mech., Aalto Univ., Puumiehenkuja 5 A, Espoo 02150, Finland, kiran.sahu@aalto.fi)

In this paper, the active control of harmonic sound transmitted through soft-cored sandwich panels into a rectangular enclosure is studied. As it has already been shown that in the low frequency region, the noise transmission through soft-cored sandwich panels mainly occurs due to flexural and dilatational modes [Rimas Vaicaitis, NASA Technical Note, NASA TN D-8516, 1977]; therefore, in this study, volume velocity cancellation control strategy is used to control these modes, and achieve sound attenuation in a broad frequency range. Point force and uniformly distributed force actuators are used as the secondary source to cancel the volume velocity of the inner surface of the sandwich panel which is open to cavity. Cancelling the net volume velocity of this is compared not only in terms of the reduction in sound power in the enclosure but also in terms of the plate velocities. Numerical studies indicate that the active control method controls both the flexural and dilatational modes and therefore, attenuates significant amount of sound power inside the cavity irrespective of the isotropic loss factors of the viscoelastic core. Also a finite element study has been done in the commercially available COMSOLMultiphysics software to compare with the analytical result.

**4:10**

**4pNS10. Effect of modeling errors on virtual sensing systems for active noise control.** Luis Vicente (Aragon Inst. of Eng. Res., Univ. of Zaragoza, Maria de Luna, 1, Zaragoza E50018, Spain, lvicente@unizar.es)

In the active noise control literature, there are a number of algorithms based on the virtual sensing approach. In all of them, the aim is to control the noise at a position apart from the physical sensors, by somehow estimating the actual signal at that point. The benefit of such an arrangement is evident. However, the practical difficulties found when trying to achieve a properly working virtual sensing system surely are the main reason for a reduced number of successful applications reported. In this paper, we analyze and quantify those difficulties for several virtual sensing algorithms. All of them are based on some modeling that needs to be made previous to the actual control. We focus on the effect that errors on these models have on the stability of the whole system, the cancellation capability and the convergence rate. We check that the sensitivity of the cancellation capability to modeling errors is much higher in the virtual sensing case, when compared to that of systems with physical sensors on the desired points of cancellation.

**4:25**

**4pNS11. Current developments in practical active damping systems for vibration-isolated platforms.** Vyacheslav Ryaboy (Newport Corp., 1791 Deere Ave., Irvine, CA 92606, vyacheslav.ryaboy@newport.com)

The paper discusses recent developments in practical implementation of the active damping solution for vibration isolated platforms (optical tables) known as Smart Table. The system implements a decentralized velocity feedback with sensors and actuators integrated into the platform. It proved effective in creating vibration-free environments for sensitive experiments and precision manufacturing processes in life sciences, nanotechnologies and other areas. The paper describes expansion of the technology to larger platforms characterized by lower resonance frequencies. The technical difficulties related to interference between the resonance properties of the structure and the resonance of the electromagnetic actuator (see Elliott and Baumann, J. Acoust. Soc. Am. **121**(5), 2007) were addressed successfully. Other developments required by expansion of the area of applications included introduction of dampers acting in all directions, as well as a portable modular version of the active dampers.

**4:40**

**4pNS12. Development of radiation mode shapes for cylindrical shells.** Pegah Aslani, Scott Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N203 ESC, Provo, UT 84602-4673, pegah.aslani@gmail.com), and Jonathan Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

For many acoustical applications, it is desirable to evaluate the radiated power. About two decades ago, a set of formulations were developed to represent the acoustic radiation in terms of radiation mode shapes. A convenient method for determining these radiation modes involves representing the radiating structure as a set of elementary simple radiators, from which the radiation can be decomposed into the set of orthogonal radiation modes. Radiation mode shapes are very useful not only for calculating the power, but also to determine which modes are the most efficient radiators. This generally allows one to achieve a rather accurate estimate of the radiated power by including only a relatively small number of the most efficient radiating modes. This concept has significant implications for an efficient strategy for implementing an active noise control system. Previous work reported in the literature has primarily focused on evaluating the radiation mode shapes of flat structures, such as beams and plates. There has not been as much reported on the radiation mode shapes for cylindrical shells. This paper focuses on implementing these concepts to determine the radiation mode shapes of cylindrical shells and using them to determine the radiated acoustic power.

**4:55**

**4pNS13. Echo removal in tubular acoustic systems: Passive and active techniques.** Keir H. Groves and Barry Lennox (Elec. and Electron. Eng., The Univ. of Manchester, Sackville St. Bldg., Granby row, Manchester M17AY, United Kingdom, keir.groves@manchester.ac.uk)

Acoustic pulse reflectometry (APR) has been shown to be a very capable means of identifying features in tubular objects. APR systems excite a test object with a sound wave and listen for reflections, indicating the presence of features in the test object. An undesirable effect of this process is that the returning sound wave is re-reflected by the loudspeaker and re-enters the system. This paper presents two complimentary techniques that may be used to remove unwanted echoes in APR systems. The first approach uses two axially separated microphones to separate forward and backward propagating waves. This passive technique is shown to be highly capable of cancelling undesired echoes in the system. The second approach actively cancels unwanted echoes by introducing a phase inverted version of the wave that is incident on the loudspeaker. The active cancellation operates in real-time using the measured backwards propagating wave. As a consequence of the proposed techniques, the effectiveness of APR when applied to detecting features within tubular systems is improved considerably. The empirical results presented at the conference will demonstrate that corrosion effects, such as holes and pits, located in short lengths of pipes, can be detected clearly within seconds.

**5:10**

**4pNS14. The effects of the orifice plate structure on the aerodynamic noise in the high parameter pressure reducing valve.** Lin Wei and Zhijiang Jin (Inst. of Chemical Machinery and Process Equipment, No. 38, Zheda Rd., Hangzhou, Zhejiang 310007, China, linweily@163.com)

The high velocity steam flow in the high pressure reducing valve can cause loud noise, which is harmful to the operators and the relevant devices. The orifice plate is used to reduce the noise in the pressure reducing valve. The main objective is to study the relationship between the orifice plate structure and the aerodynamic noise in the high pressure reducing valve. Based on computational fluid dynamics hybrid approach was used to simulate the flow and the acoustic field in the valve. The thickness of the plate, the length between the plate and the plug, the bore diameter and its distribution were changed to analyze their effects on aerodynamic noise.

## Session 4pPA

## Physical Acoustics: Advances in Infrasound Research II

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

John Heffington, Cochair
*NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677*

### *Invited Paper*

**1:30**

**4pPA1. Partitioning of seismo-acoustic motions for near-surface explosions and yield estimation.** Arthur Rodgers (LLNL, 7000 East Ave., L-046, Livermore, CA 94551, rodgers7@llnl.gov)

Explosions near the Earth's surface excite both atmospheric overpressure and seismic ground motions. The amplitudes of air-blast (and hence acoustic/infrasound) overpressures and seismic motions depend on the explosive yield as well as the height-of-burst (HOB, for above ground emplacement) or depth-of-burial (DOB, for buried emplacement). We present analysis of air-blast overpressures and seismic motions with the goal of developing methods for robust yield estimation for near-surface blasts. Our investigations are based on the HUMBLE REDWOOD set of chemical high-explosive tests at Kirkland Air Force Base in Albuquerque, NM. We find that the air-blast positive phase impulse and seismic P-wave zero-to-peak displacement amplitude are robust estimators of yield. An empirical model for the amplitudes as a function of yield, range and HOB/DOB is presented and allows estimation of yield and HOB/DOB given a set of air-blast and seismic measurements. We find that yield and HOB/DOB can be estimated simultaneously by combining air-blast and seismic measurements. Strong trade-offs between the amplitudes and the yield and HOB/DOB for a single measurement type inhibit accurate estimates. However, simultaneous inversion of both overpressure and seismic measurements improve estimates, justifying combined seismo-acoustic analysis.

### *Contributed Papers*

**1:50**

**4pPA2. Observations on geomagnetic auroral infrasound waves 2003—2013.** Justin J. Oldham, Charles R. Wilson, John V. Olson, Curt Szuberla, and Hans Nielsen (Phys., Univ. of Alaska Fairbanks Geophysical Inst., PO Box 750972, Fairbanks, AK 99775, joldham6@alaska.edu)

Persistent, high trace velocity infrasound activity, associated with auroral events, has been routinely observed from the CTBT/IMS I53US infrasound station in Fairbanks. Comparisons of the infrasound data with data from the Geophysical Institute Magnetometer Array, the Poker Digital All-Sky Camera, and historic data from the Poker Flat Imaging Riometer show that the observed infrasound is correlated with periods of heightened geomagnetic activity and is produced in the lower Ionosphere. With the infrasound array operating near-continuously now for roughly one full period of the solar cycle, we have systematically isolated all such geomagnetic auroral infrasound events to form a data set suitable for statistical analysis. We note a relationship between the occurrence of geomagnetic infrasound waves (GAIW) and the recovery phase of geomagnetic storms when the geomagnetic H component has a peak-to-peak amplitude of ~1500 gamma during the local time period from 5 to 10 h at the CIGO Magnetic Observatory in Fairbanks. During this time interval I53US is under the auroral oval when there are large pulsating aurora events that produce infrasonic waves. These observations restrict the apparent source geometry and generating phenomena of the infrasound, as well as providing a basis for comparison with idealized models of GAIW generation.

**2:05**

**4pPA3. Pneumatic infrasound source: Model experiment and theory.** Justin D. Gorhum, Thomas G. Muir, Charles M. Slack III, Timothy W. Hawkins, Yurii A. Ilinskii, and Mark F. Hamilton (Appl. Res. Lab., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713, jgorhum@arlut.utexas.edu)

In a previous presentation [J. Acoust. Soc. Am. **133**, 3327 (2013)], an experimental model study of a pneumatic infrasound source that utilizes the pulsation of compressed air was discussed. The present paper discusses new measurements and theoretical modeling efforts that are currently underway. Measurements of the source level, directivity patterns, propagation loss, and frequency response are presented and analyzed. Acoustic and aerodynamic models are presented and discussed with a focus on modeling and predicting nearfield system performance using multipole (monopole, dipole, and quadrupole) representations of the sound source. Measurement techniques and engineering considerations are addressed, as are physical interpretations of the process. [Work supported by ARL:UT Austin.]

**2:20**

**4pPA4. Pneumatic infrasound source: Expanded model development and tests.** Thomas Muir, Charles M. Slack, Justin D. Gorhum, and Timothy M. Hawkins (ARL UT Austin, P.O. Box 8029, Austin, TX 78713, tgmuir@earthlink.net)

A model experiment discussed in a companion paper is expanded through the engineering development of a larger scale system to provide concept evaluation of portable infrasound generation, for calibration and

tests of receiver array stations. This system utilizes an industrial compressor producing 350 cubic feet per minute of air at pressures up to 150 pounds per square inch, which is stored in two 500 gallon tanks. Air streams from each tank are released through two synchronized, rotating, 2 in. diameter ball valves, producing modulated pulse jets into the atmosphere, which then create infrasonic tone bursts. Measurements on the propagation and frequency response of infrasound so produced are compared and modeled with a view toward assessment of practical utility. [Work supported by Applied Research Laboratories, University of Texas at Austin.]

**2:35**

**4pPA5. Acoustic signals and directivity for explosive sources in complex environments.** Roger M. Waxler (NCPA, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu), Doru Velea (SAIC, Reston, VA), Jessie Bonner (Weston Geophysical Corp., Boston, MA), and Carrick Talmadge (NCPA, Univ. of Mississippi, University, MS)

Much work has gone into characterizing the blast wave, and ultimate acoustic pulse, produced by an explosion in flat, open land. Recently, an experiment was performed to study signals produced by explosions in more complex environments, both above and below ground and in the vicinity of mountainous terrain. Explosive charges, ranging in weight from 200 to 2000 lbs, were detonated in a variety of configurations in and around tubes and culverts as well as buried in alluvium and limestone. A large number of acoustic sensors were deployed to capture the directivity of the signals in the near-field and to characterize the propagation of the signal to the far field. Significant directivity was observed in the near field signals from many of the shots. The influences of both meteorology and topography were evident.

**2:50**

**4pPA6. Infrasound from buried seismic sources in the presence of surface topography.** Arthur Rodgers (LLNL, 7000 East Ave., L-046, Livermore, CA 94551, rodgers7@llnl.gov)

Buried seismic sources (such as explosions and earthquakes) can generate acoustic motions in the atmosphere through coupling along the solid-fluid boundary. Infrasound overpressures from such sources have been computed using the Rayleigh Integral where acceleration time-histories along the boundary are inversely weighted by distance, delayed by travel time and summed at an observation point in the far-field. Typically, these calculations assume the Earth's surface is flat; however, topography can result in variations of the overpressure signals due to amplitude differences at the surface and phase differences along the direction of propagation. This study considers the Rayleigh Integral to compute far-field overpressure using seismic ground motion simulations that include accurate representation of surface topography. Through a series of numerical experiments we attempt to quantify the effect of surface topography on overpressure signals.

**3:05**

**4pPA7. An empirical study of acoustic/infrasonic source and propagation effects using a large dataset of explosions.** Emily A. Morton (Geophys., Los Alamos National Lab., 4129 S Meadows Rd., Apt. 2121, Santa Fe, NM 87507, emorton@lanl.gov) and Stephen J. Arrowsmith (Geophys., Los Alamos National Lab., Los Alamos, NM)

In May 2013 we performed a series of seventy explosion tests, varying the mass, shape, and height of the explosives. Shots were comprised of 11.6 kg, 4.9 kg, and 1.7 kg cylinders and 14.9 kg spheres, all of Comp-B. Explosive heights varied between 4, 2, 1, and 1/2 m above the surface, at the surface, and buried 1 m below the surface. Explosives above the surface were suspended by rope between two concrete pillars. In addition, ground surfaces were altered between dry sand, chicken wire, and concrete blocks. We monitored the explosions on 13 acoustic stations. Four temporary stations were deployed surrounding the shot site at less than 1 km distance. Eight additional stations were at distances of 1 to less than 9 km, and one at ~23 km from the shot site, 4 of which were temporary stations, and 5 are part of the Los Alamos Seismo-acoustic Network. We report on a detailed analysis of signal differences related to explosive and meteorological variations. The large quantity of data from repeating shots enables us to formally characterize the relative importance of source and path variations.

**3:20**

**4pPA8. Comparison of primary and secondary calibrations of infrasound microphones from 0.01 to 20 Hz.** Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

Secondary calibration of microphones at infrasonic frequencies by comparison to a reference pressure transducer in a piston-driven chamber is straightforward as long as the two transducers can be located much closer than a wavelength or a correction for their separation can be determined accurately. If the response of the reference transducer is flat to zero frequency, the reference can be calibrated statically. For comparison calibration, the uncertainty is dominated by the uncertainty in the reference. In this investigation, a calibration chamber that is normally used for comparison calibration has been analyzed for primary calibration. In the primary mode, the calibration depends on chamber dimensions, piston displacement, temperature, barometric pressure, leak rate, and a thermo-viscous acoustic model. The primary and secondary calibrations are performed simultaneously; however, the two calibration modes produce almost entirely independent response estimates of both magnitude and phase. The calibrations extend well below the nominal low-frequency roll-off of the microphone and allow identification of the characteristics of the pressure-equalization leak. In addition to the linear analysis, the effects of nonlinearity and convection are explored.

**3:35**

**4pPA9. Lightning characterization through acoustic measurements.** Louis-Jonardan Gallin (DAM, DIF, CEA, CEA/DAM Ile de France, Arpajon 91297, France, gallin@dalembert.upmc.fr), Mathieu Rénier, Éric Gaudard (Institut Jean le Rond d'Alembert, UPMC, Paris, France), Thomas Farges (Institut Jean le Rond d'Alembert, UPMC, Arpajon, France), Régis Marchiano (Institut Jean le Rond d'Alembert, UPMC, Paris, France), and François Coulouvrat (Institut Jean le Rond d'Alembert, CNRS, Paris, France)

Lightning generated acoustic shock waves are the most frequent natural explosions: they are good candidates to probe meteorological local properties of the acoustic propagation medium over distances of less than 100 km. The goal of the Ph.D. is to study the transformation the thunder undergoes (amplitude, spectrum) during its travel from the lightning channel towards a detector (microphone, microbarograph), the work is based on two complementary approaches: first the Flhoward software (UPMC) designed to simulate the propagation of acoustic shock waves through a realistic atmosphere model (including temperature gradients, rigid ground, and winds) will help us studying the traveling waveforms. And in second the analysis of the acoustic records (audible and infrasounds) obtained during the PEACH campaign (Autumn 2012) will provide data to which simulations will be confronted.

## Session 4pPP

# Psychological and Physiological Acoustics: The Ear Club: Honoring Ervin R. Hafter and His Contributions to the Study of Binaural Processing and Auditory Cognition II

Brent Edwards, Chair
*Starkey Hearing Res. Ctr., 2150 Shattuck Ave., Berkeley, CA 94704*

### *Invited Papers*

**2:00**

**4pPP1. Lateralization of simulated sources and echoes on the basis of interaural differences of level.** Raymond H. Dye, Jacquelyn P. Hill, Leslie M. Ryan, Alexander E. Cupler, and Kevin M. Bannon (Psych., Loyola Univ. Chicago, 1032 W. Sheridan Rd., Chicago, IL 60201, rdye@luc.edu)

This experiment assessed the relative weights given to source and echo pulses lateralized on the basis of interaural differences of level (IDLs). Separate conditions were run in which the to-be-judged target was the first (source) or second (echo) pulse. Each trial consisted of two intervals; the first presented a 3000-Hz diotic pulse that marked the intracranial midline and the pitch of the target frequency. The second presented the sequence of a source followed by an echo. Target frequency was always 3000 Hz, while the non-target pulse was presented at 1500, 3000, or 5000 Hz. Delays between the source and echo were varied from 8 to 128 ms. IDL's were chosen for both pulses from Gaussian distributions with $\mu = 0$ dB and $\sigma = 4$ dB. Dependent variables included normalized target weight, proportion correct, and the proportion of responses predicted from the weights. Although target weight and proportion correct generally increased with increasing non-target frequency and echo delay for both target conditions, the effects were always larger when the echo served as the target. The superiority of performance when judging echoes vs sources will be discussed in terms of recency effects in binaural hearing.

**2:20**

**4pPP2. Within- and across-channel integration of information for the precedence effect.** Bernhard U. Seeber (Audio Information Processing, Technische Universität München, Arcisstrasse 21, Munich 80333, Germany, seeber@tum.de)

Erv Hafter has with his team investigated the spectral and temporal integration of binaural information to learn about binaural hearing in situations with multiple sounds and reflections. For these studies, the Simulated Open Field Environment (SOFE), a loudspeaker setup in an anechoic chamber, was created. Beginning with an overview of the SOFE, I will present results on the spectral density and bandwidth of long-duration complex tones needed for the precedence effect to occur, the ability to correctly locate a sound in the presence its delayed copy. The hypothesis is that the precedence effect cannot be evoked with a single low frequency tone, because the addition of its delayed copy alters the interaural phase and thus its location. A larger bandwidth is needed to stabilize localization at the lead either through integrating binaural information across frequency or through extracting information from the temporal envelope. Results show that a stable precedence effect could not be obtained at or below 1 Bark bandwidth, and that at least two tones per critical band over 2 Bark are required. The echo threshold increases with increasing bandwidth or spectral density, suggesting that within and across-channel information is combined.

**2:40**

**4pPP3. Where am I, where is the sound source?** Yost A. William, Xuan Zhong, Anbar Najam (Speech and Hearing Sci., ASU, PO Box 870102, Tempe, AZ 85257, william.yost@asu.edu),

Locating sound sources in the everyday world often involves listeners and sources who change location. Since sound source localization cues are relative changing when the listener or source moves, veridical and accurate sound source localization when the listener moves requires that the auditory brain "knows" where the listener is in 3-dimensional space. Experiments were conducted in a sound-deadened room with 36 loudspeakers located on a 5-foot radius sphere and a rotating chair. The listener was rotated in accelerating, decelerating, and constant velocity conditions and was either sighted (eyes open) or blind folded (eyes closed). The sound source was either fixed at one location, or the sound (100-ms, broadband noise) changed position from one loudspeaker to the next (along a 24-loudspeaker azimuth circle) in an accelerating, decelerating, or constant velocity manner. In all cases but one, listeners were able to perceive the loudspeaker presenting the sound in the same way they did when the listener was stationary. When the eyes were closed (no visual cues) and the chair was rotated at constant velocity (no semicircular canal vestibular cues), listeners badly misperceived sound source locations. The results indicate that veridical sound source localization requires visual and/or vestibular information.

**3:00–3:15 Break**

**3:15**

**4pPP4. Revisiting the loudness of sounds with asymmetric attack and decay.** Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

Stecker and Hafter [J. Acoust. Soc. Am. **107**, 3358–3368 (2000)] compared the loudness of sounds whose envelopes had a fast attack and a slow decay (designated F-S) and a slow attack and a fast decay (designated S-F). They found that, for sinusoidal and broadband noise carriers, S-F stimuli were louder than F-S stimuli of equal energy. They argued that this effect could not be explained by current models of loudness and that the loudness effect may be related to the parsing of auditory input into direct and reverberant sound. Subsequent work has shown that the differences in loudness between F-S and S-F stimuli can be partially accounted for by the loudness model of Glasberg and Moore [J. Audio Eng. Soc. **50**, 331–342 (2002)], which incorporates a form of temporal averaging that is asymmetric in time. However, the model does not account for the context effect found by Stecker and Hafter. The largest differences in loudness between F- S and S-F stimuli occurred after pre-exposure to a F-S stimulus. This may happen because, when successive sounds have similarly slow decays, the decaying part is attributed to room reverberation and contributes less to loudness.

**3:35**

**4pPP5. Attention and the refinement of auditory expectations.** Psyche Loui (Psych. and Neurosci. and Behavior, Wesleyan Univ., 12 Chestnut St., Cambridge, Massachusetts 02139, ploui@wesleyan.edu)

Although traditional approaches in psychoacoustics emphasize bottom-up processes of hearing, a lasting approach of the Hafter lab has been to merge the bottom-up view with top-down influence of cognitive and training-related factors. Much of this is embodied in a research program on auditory attention: the listener's ability to extract relevant features of the auditory scene (Hafter *et al*., 2007). I joined the Hafter lab with interests in auditory attention and music perception. In work conducted in the lab we observed effects of training on attentive processing of musical harmony—musicians are slower to respond to musically unexpected harmonies but faster to respond to expected harmonies, suggesting that long-term training refines the expectations that are built up from lifelong exposure to music in one's culture. Armed with this knowledge, we further asked if these expectations could be learned. Using a non-Western musical scale, we showed rapid learning of perceptual patterns, which can be modeled as a reduction in uncertainty and an increase in correlation with the auditory environment. In subsequent work we combined electrophysiology, neuropsychology, and neuroimaging to show that the learning mechanisms that sharpen auditory expectations are rapid, flexible, and depend on neural connectivity that also subserves linguistic processes.

**3:55**

**4pPP6. Attention and effort during speech processing.** Anastasios Sarampalis (Univ. of Groningen, Grote Kruisstraat 2/1, Groningen 9712TS, Netherlands, a.sarampalis@rug.nl)

The concepts of attention and effort are not new in auditory science, yet it is only recently that we have started systematically studying their involvement in speech processing. The task of deciphering speech can vary in its cognitive demands, depending on a number of factors, such as sound quality, the state of the auditory and cognitive systems, room acoustics, and the semantic complexity of the signal itself. Understanding these interactions does not only shed light on the functions supporting speech processing, but is also critical when it comes to evaluating new hearing aids and cochlear implant strategies. This presentation will describe work that is either based on Erv Hafter's ideas while I was at UC Berkeley or inspired by discussions with him in subsequent years. Its central theme is the measuring of listening effort and its implications to digital signal processing, cochlear implants, aging, or understanding non-native languages.

**4:15**

**4pPP7. Spatial release of the cognitive effort of understanding speech in multi-talker environments.** Sridhar Kalluri, Jing Xia, Nazanin Nooraei, and Brent Edwards (Starkey Hearing Res. Ctr., Starkey Hearing Technologies, 2150 Shattuck Ave., Ste. 408, Berkeley, CA 94704, sridhar_kalluri@starkey.com)

Ervin Hafter was prescient in recognizing the need for taking into account top-down cognitive processing for understanding the interaction between auditory perception and cognition. His insight, that signal processing such as noise reduction may modify cognitive demands without changing auditory performance, led to a seminal study in collaboration with the Starkey Hearing Research Center which showed that hearing aid technology can reduce the cognitive effort of understanding noisy speech. Inspired by Erv's insight, we are studying if increasing the spatial separation between competing talkers reduces the cognitive effort needed to listen in multi-talker environments. Following the lead of Erv's seminal study, performance on a simultaneous secondary task, in our case visual tracking, is a measure of the cognitive effort consumed by the primary task of understanding target speech. Our results show that spatial separation can reduce cognitive effort even when it does not give further improvement in speech intelligibility over existing segregation cues. These results suggest that a measure of cognitive effort is useful for assessing the benefit of hearing technology that improves spatial segregation. This is an important finding because the measure addresses benefit along a dimension that is not captured in standard assessment of speech reception performance.

**4:35–4:45 Concluding Remarks**

## Session 4pSA

## Structural Acoustics and Vibration and Architectural Acoustics: Structural and Acoustic Response Due to Impulsive Excitation

Marcel Remillieux, Chair
*Los Alamos National Lab., Geophysics Group (EES-17), M.S.: D446, Los Alamos, NM 87545*

**Chair's Introduction—1:30**

### *Invited Papers*

**1:35**

**4pSA1. Inelastic deformation and failure of partially strengthened profiled blast walls.** Arash Soleiman-Fallah, Ebuka Nwankwo (Civil Eng., Imperial College London, Skempton Bldg., Exhibition Rd., South Kensington Campus, London SW18 4GR, United Kingdom, as3@imperial.ac.uk), Genevieve S. Langdon (Mech. Eng., Univ. of Cape Town, Cape Town, South Africa), and Luke A. Louca (Civil Eng., Imperial College London, London, United Kingdom)

Blast walls that separate the potentially hazardous regions of the topside on an offshore platform were designed to resist lower loads than those envisaged today thus it is desirable to upgrade their blast resistance in a cost-effective and non-intrusive manner. One proposal is to retrofit the existing blast walls partially with centrally located composite patches. This study presents an assessment tool, which provides understanding of the effect of a composite patch on the blast resistance of blast walls. Numerical simulations of a proposed retrofitted wall are performed to gain insight into the failure progression of the wall *ab initio*. Damage in the composite patch was considered, and the numerical simulations showed that fiber fracture did not occur thus there was no significant loss of in-plane stiffness and strength. Based on these observations, the rapid assessment tool, analytically formulated to incorporate the effect of the composite patch which strengthens the wall and moves the plastic hinge locations away from the wall centre to the composite-steel edge, is deemed a suitable tool. The assessment tool and the numerical simulations are partially validated by the experimental results. The tool runs quickly and provides reasonable accurate predictions for the deformation response of the walls.

**1:55**

**4pSA2. Predicting the response of structures to transient shock loading.** Mauro Caresta, Robin S. Langley, and Jim Woodhouse (Eng., Univ. of Cambridge, Trumpington St., Cambridge CB21PZ, United Kingdom, maurorestaca@yahoo.it)

This work concerns the prediction of the response of an uncertain structure to a load of short duration. Assuming an ensemble of structures with small random variations about a nominal form, a mean impulse response can be found using only the modal density of the structure. The mean impulse response turns out to be the same as the response of an infinite structure: the response is calculated by taking into account the direct field only, without reflections. Considering the short duration of an impulsive loading, the approach is reasonable before the effect of the reverberant field becomes important. The convolution between the mean impulse response and the shock loading is solved in discrete time to calculate the response at the driving point and at remote points. Experimental and numerical examples are presented to validate the theory presented for simple structures such as beams, plates, and cylinders.

**2:15**

**4pSA3. Characterization of a spark source focused by an ellipsoidal reflector.** Xiaowei Dai, Yi-Te Tsai (Civil, Architectural, and Environ. Eng., The Univ. of Texas at Austin, 301 E. Dean Keeton St., M.S. C1747, Austin, TX 78712, jyzhu@mail.utexas.edu), Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Jinying Zhu (Civil, Architectural, and Environ. Eng., The Univ. of Texas at Austin, Austin, TX)

Air-coupled ultrasonic non-destructive testing (NDT) provides an ideal solution for rapid scanning of large specimens. Unfortunately, despite decades of research, many challenges remain to render air-coupled ultrasonic methods a broadly effective sensing modality for high impedance materials due to low energy transmission between air and the solids being inspected. In this paper, we present experimental results and theoretical analysis of an electrical spark source focused by an ellipsoidal reflector. This acoustic source, which generates a short duration, high amplitude signal in air, is of high interest for air-coupled NDT for high impedance materials and has been shown to excite wave motion in concrete without contact. Theoretical modeling using weak shock theory and the KZK equation is used to predict the temporal and spatial features of the pressure field in the region of the geometric focus. We also present a series of experimental studies that characterize the spark generated acoustic wave in both free-field and the focused conditions. The bandwidth and directivity of the focused spark source are shown to be adjustable by changing the spark gap size and the reflector geometry. Finally, experimental results from three reflectors made of different material and geometries are presented.

**2:35**

**4pSA4. Dynamic acousto-elasticity in Berea sandstone: Influence of the strain rate.** Jacques Riviere (EES-17, Los Alamos National Lab., MS D446, Los Alamos, NM 87545, riviere_jacques@yahoo.fr), Thibault Candela, Marco Scuderi, Chris Marone (GeoSci., Penn State Univ., University Park, PA), Robert Guyer, and Paul A. Johnson (EES-17, Los Alamos National Lab., Los Alamos, NM)

In comparison with standard nonlinear ultrasonic methods such as frequency mixing or resonance based measurements that allow one to extract average, bulk variations of modulus and attenuation versus strain level, dynamic acousto-elasticity (DAE) allows to obtain the elastic behavior over the entire dynamic cycle, detailing the full nonlinear behavior under tension and compression, including hysteresis and memory effects. To improve our understanding of these phenomena, this work aims at comparing static and dynamic acousto-elasticity to evaluate the influence of strain rate. To this purpose, we perform acousto-elasticity on a sample of Berea sandstone and a glass beads pack, oscillating them from 0.001 to 10 Hz. These results are then compared to DAE measurements made in the kHz range. We observe that the average decrease in modulus increases with frequency, meaning that conditioning effects are higher at high strain rate, when relaxation characteristic time is higher than the oscillation period. This result, together with previous quasi-static measurements (Claytor *et al.*, GRL 2009) showing that the hysteretic behavior disappears when the protocol is performed at a very low strain-rate, confirms that a rate dependent nonlinear elastic model has to considered for a more complete description (Gusev *et al.*, PRB 2004).

**2:55–3:10 Break**

**3:10**

**4pSA5. Semi-analytical study of interfacial stresses in adhesively bonded single lap joints subject to transverse shock loading.** Ebuka Nwankwo, Arash Soleiman-Fallah, and Luke A. Louca (Civil Eng., Imperial College London, Skempton Bldg., Exhibition Rd., South Kensington Campus, London sw7 2az, United Kingdom, en208@imperial.ac.uk)

Debonding in adhesively bonded lap joints is a detrimental failure mode contingent upon the level of stresses developed in the adhesive. A semi-analytical model is developed to estimate the peel and shear stresses in an isotropic elastic adhesive in a single lap joint subjected to transverse shock loads. The proposed semi-analytical model is an extension of existing mathematical models to study the coupled transverse and longitudinal vibrations of a bonded lap joint system. The adhesive is modeled as an isotropic material in ABAQUS. The interfacial stresses obtained by finite element simulations were used to validate the analytical model. The maximum peel and shear stresses predicted by the analytical model in the adhesive were found to correlate well with the maximum stresses predicted by the corresponding numerical models. The peel stresses in the adhesive were found to be higher than shear stresses, a result which is consistent with intuition for transversally pulse loaded joints. The semi-analytical model is able to predict the maxim stresses in the edges where debonding initiates due to the highly asymmetrical stress distribution as observed in the finite element simulations and experiment. The stress distribution under uniformly distributed transverse pulse loading was observed to be similarly asymmetric.

## *Contributed Papers*

**3:30**

**4pSA6. Structural infrasound from a barge collision with the Mississippi River Bridge.** Anna M. Miller, Richard D. Costley, Henry Diaz-Alvarez, Mihan H. McKenna, and Christopher P. Simpson (GeoTech. & Structures Lab., US Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, anna.m.millter@usace.army.mil)

The Mississippi River Bridge in Vicksburg MS is a 7 span cantilever bridge 3389 feet long by 68.5 ft wide and is part of the Interstate-20 corridor. On 23 March 2011 around 1:30 pm, a barge moving downstream struck a pier of the bridge. Infrasound stations located at the Waterways Experiment Station (WES) detected the impact. There are indications from the infrasound signatures that infrasound was radiated from the bridge and also from disturbances on the surface of the water (waves or eddies) that resulted from the collision. Finite Element (FE) models of the bridge and pier were developed to simulate the response of the bridge due to the barge impact. It was possible to identify those portions of the infrasound signature produced by vibration of the bridge deck. A synopsis of the accident will be presented along with the recorded infrasound signatures. Results from the dynamic structural model of the bridge will be discussed and related to the infrasound signature.

**3:45**

**4pSA7. A hybrid numerical model for the exterior-to-interior transmission of impulsive sound through three-dimensional, thin-walled elastic structures.** Marcel C. Remillieux (Geophys. Group (EES-17), Los Alamos National Lab., Los Alamos, NM 87545, mcr1@lanl.gov), Stephanie M. Pasareanu (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA), and U. Peter Svensson (Dept. of Electronics and Telecommunications, Norwegian Univ. of Sci. and Technol., Trondheim, Norway)

Exterior propagation of impulsive sound and its transmission through three-dimensional, thin-walled elastic structures, into enclosed cavities, are investigated numerically in the framework of linear dynamics. A hybrid model was developed in the time domain by combining the advantages of two existing numerical tools: (i) exterior sound propagation and induced structural (façade) loading are computed using the image-source method for the reflected field (specular reflections) combined with an extension of the Biot-Tolstoy-Medwin method for the diffracted field, (ii) the fully coupled vibro-acoustic response of the interior fluid-structure system is computed using a truncated modal-decomposition approach. In the model for exterior sound propagation, it is assumed that all surfaces are acoustically rigid. Since coupling between the structure and the exterior fluid is not enforced, the model is applicable to the case of a light exterior fluid and arbitrary interior fluid(s). The structural modes are computed with the finite element method using shell elements. Acoustic modes are computed analytically assuming acoustically rigid boundaries and rectangular geometries of the enclosed cavities. This model is verified against finite-element solutions computed with a commercial software package for the cases of rectangular structures containing one and two cavities, respectively.

**4:00**

**4pSA8. Theoretical and experimental analysis of shock isolation using non linear stiffness.** Diego Ledezma, Jose de Jesus Villalobos-Luna (Facultad de Ingenieria Mecanica y Electrica, Universidad Autonoma de Nuevo Leon, Av Universidad sn, San Nicolas de los Garza, Nuevo Leon 66456, Mexico, diego.ledezmard@uanl.edu.mx), Neil Ferguson (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom), and Michael Brennan (Departamento de Engenharia Mecanica, UNESP, Ilha Solteira, Brazil)

Shock vibration is a common problem involving large forces and accelerations, usually resulting in nonlinear behavior. Normally shock isolation systems are modeled after linear passive stiffness elements intended to absorb the energy from the shock, and viscous damping in order to dissipate the energy. An experimental system with low dynamic stiffness is proposed and presented in this work, using a combination of positive stiffness and negative stiffness given by magnetic forces. The experimental prototype is

based on a theoretical model involving a cubic restoring force. The results presented shown how such an isolator provides improved shock isolation, pointing out advantages and disadvantages.

**4:15**

**4pSA9. A study on structural vibration of washing machine with gyroscope.** Gyu Sung Na, Young Jin Park, Yoon Sik Park (Mech. Eng., KAIST, 291 Daehak-ro, Yuseong-gu, Daejeon 305-701, South Korea, joycap01@kaist.ac.kr), and Jeong Hoon Kang (Digital Appliances, Samsung Electronics, Suwon, South Korea)

This paper is proposed about reducing the transient vibration of drum type washing machine. The vibration of washing machine is caused by unbalanced cloths in high spinning drum, and the displacement of tub is maximized at transient range about 3 Hz (180 rpm). The dynamic model of washing machine is include a diaphragm. In this study, the displacement of tub is decreased by using gyroscope system. Multibody dynamic model of washing machine include gyroscope is designed and the vibration of tub have been reduced by the gyroscope system.

THURSDAY AFTERNOON, 5 DECEMBER 2013 PLAZA B, 1:30 P.M. TO 5:00 P.M.

# Session 4pSCa

## Speech Communication: Language Description

Natasha L. Warner, Chair
*Dept. of Linguist., Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028*

### *Contributed Papers*

**1:30**

**4pSCa1. Best practices in measuring vowel merger.** Jennifer Nycz (Dept. of Linguist., Georgetown Univ., 1437 37th St. NW, Washington, DC 20057, jn621@georgetown.edu) and Lauren Hall-Lew (Linguist. and English Lang., The Univ. of Edinburgh, Edinburgh, United Kingdom)

Vowel mergers are some of the most well-studied sound change phenomena, particularly in varieties of English. But although sociolinguists, dialectologists, and phoneticians are all interested in providing accurate and precise descriptions of an individual speaker's participation in a near-merger (or near-split), the methods for doing so vary widely, especially for researchers analyzing naturalistic corpora. In this paper, we consider four current methodological approaches to representing and assessing vowel distance and overlap: Euclidean distances between averages, Pillai-Bartlett trace (Hay *et al.*, 2006), mixed effects regression modeling (Nycz 2013), and the spectral overlap assessment metric (Wassink 2006). We compare the advantages and disadvantages of each by applying all four methods to three separate data sets. These represent low vowel realizations by speakers from three different studies of English variation: one undergoing merger (COT and CAUGHT in San Francisco, California), one undergoing split, for the same contrast (COT and CAUGHT among Canadians in New York City), and one undergoing split, but for a different contrast (TRAP and BATH among Scots in England). By comparing the similarities and differences between the data sets themselves, as well as the differing analytic motivations for quantifying speaker-specific vowel overlap, we conclude with practical recommendations.

**1:45**

**4pSCa2. The use of high rise terminals in Southern Californian English.** Amanda Ritchart (Linguisit., UCSD, 1 Miramar St. #929004, La Jolla, CA 92092, aritchart@ucsd.edu) and Amalia Arvaniti (English Lang. & Linguist., Univ. of Kent, Kent, United Kingdom)

This study investigates High Rise Terminals (HRTs), i.e., utterance-final rising pitch movements, as used in Southern Californian English (SoCalE), examining the phonetics and phonology of HRTs and their relation to pragmatic functions. Twelve female and 11 male speakers were recorded during a map task and in the retelling of a sitcom scene. HRTs were coded for discourse function (statement, question, confirmation request, floor holding) based on context. The alignment of the pitch rise start was measured from the onset of the utterance's last stressed vowel, and the rise's final Hz value was recorded. In HRTs used for statements, the rise started within the stressed vowel, while in questions it started after vowel offset. Together with the low F0 on the stressed syllable, this pattern suggests that statements have a L*L-H% melody while questions have L*H-H%. Confirmation requests and floor holding were more variable in alignment. Consistent differences in pitch scaling were found in the order: questions, confirmation requests > floor holding > statements. Females used HRTs more often than males, and their HRTs showed greater pitch excursion and later alignment. In conclusion, SoCalE uses different HRT melodies than other varieties and maintains a distinction between HRTs for statements and questions.

**2:00**

**4pSCa3. Phonetic shift across narrative and quoted speech styles.** Paul De Decker (Dept. of Linguist., Memorial Univ. of Newfoundland, St. John's, NF A1B 3X8, Canada, pauldd@mun.ca)

Qualitative descriptions of speech accompanying verbs of quotation (e.g., "She was like, 'I'm not going in there!'") characterize quoted speech as a mimetic performance (Buchstaller 2003, Winter 2002) with "selective depictions" of the quotees words (Clark and Gerrig 1990, 1). The current study aims to quantify the performative and mimetic nature of quoted speech by comparing acoustic measurements of 539 vowel productions obtained through narratives of personal experience (Labov and Waletzky 1967) as told between friends. First and second formant frequencies were measured at the temporal midpoint of each vowel using PRAAT5.3 (Boersma and Weenink 2012), normalized using the BARK method and compared in a one way ANOVA in SPSS. The dependent variables were F0, F1, F2, and duration while gender of speaker and lexical set key word (Wells 1982) served as the independent variables. Results indicate that mainly female speakers showed phonetically shifted vowel quality features when moving from narrative style speech to quoting voices for characters in their stories. This specific type of phonetic alteration across speech styles is examined as a type of "speech play" (Sherzer 2002) and its role in story-telling is examined further.

**2:15**

**4pSCa4. Mon voice registers: Acoustics and laryngeal control.** Arthur S. Abramsonn, Mark K. Tiede (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, arthur.abramson@uconn.edu), and Theraphan Luangthongkum (Linguist., Chulalongkorn Univ., Bangkok, Thailand)

Mon is spoken in many villages in Thailand and Myanmar. The dialect of Ban Nakhonchum, Ratchaburi Province, Thailand, has two voice registers, modal and breathy, phonation types that, along with other phonetic properties, commonly distinguish registers. Four native speakers recorded several repetitions of 14 randomized words (seven minimal pairs) for acoustic analysis. We used a subset of these pairs for listening tests to verify the perceptual robustness of the distinction. Four speakers, three of the original ones and one new one, were also recorded using electroglottography (EGG) while repeating the word set several times. The listening tests showed the distinction to be robust. Acoustic analysis of both sets of recordings was done using the UCLA VoiceSauce program. Differences in noise component (ratio of harmonics to noise and cepstral peak prominence), spectral slope, fundamental frequency, and formant frequencies all differ across the registers. For analysis of the EGG data we used the UCLA EGGWorks program to obtain closure quotient (CQ) measures. CQ was significantly different for all four speakers with higher values for the modal register. The salience of these cues in maintaining the register distinction will be discussed. [Work supported by NIH grants and the Thailand Research Fund.]

**2:30**

**4pSCa5. The case for strident vowels.** Matthew Faytak (Linguist., Univ. of California Berkeley, 2632 San Pablo Ave., Apt. A, Berkeley, CA 94702, mf@berkeley.edu)

I present evidence for a natural class of strident vowels characterized by significant high-frequency energy caused by turbulent airflow. This turbulent airflow is not incidental to a narrow articulatory "tube," as is common for high vowels (Klatt 1975, Ohala and Solé 2010). Rather, all share an acoustic signal consistent with turbulence produced by a jet of air angled so as to strike an obstacle anterior to the jet, as seen in strident fricatives (Shadle 1990). The Mandarin words "four" [sz□] and "ten" [ʂʐ̩□] provide examples of these vowels at alveolar and retroflexed places of articulation; I provide further examples, including labiodentals, from my research on the Kom language of Cameroon. Vowels are essential for clear and reliable perception of speech, as their low spectral center of gravity, high intensity, and open articulatory configuration allow for the realization of cues to perception of neighboring, less intrinsically perceptible consonantal segments (Liberman *et al.*, 1954). Strident vowels, with their higher center of gravity, lower intensity, and consonant-like articulation, call into question the nature of this modulation, suggesting the utility of broader definitions for a sufficiently perceptible modulation in the speech signal (Kawasaki-Fukumori and Ohala 1997).

**2:45–3:00 General Discussion**

**3:00–3:30 Break**

**3:30**

**4pSCa6. Falling diphthongs have a dynamic target while rising diphthongs have two targets: Acoustics and articulation of the diphthong production in Ningbo Chinese.** Fang Hu (Inst. of Linguist., Chinese Acad. of Social Sci., 5 Jian Guo Men Nei St., Beijing 100732, China, hufang@cass.org.cn)

It is controversial whether diphthongs are phonologically vowel sequences and thus phonetically have two targets or diphthongs are phonologically vowel phonemes that contrast with monophthongs and thus phonetically have one dynamic target. Chinese dialects are generally known as having a rich inventory of diphthongs, and typically there are both falling and rising diphthongs. This paper is an acoustic and articulatory study on the diphthongs in Ningbo Chinese. The acoustic data are from 20 speakers and the lingual kinematic data are collected from 6 speakers by using EMA. The acoustic results show that both the onset and offset elements have comparable formant frequency patterns to their corresponding target citation vowels in a rising diphthong, but in a falling diphthong, only the onset element has a comparable formant frequency pattern to its corresponding target citation vowel whereas the offset element is highly variable. The articulatory results further reveal that diphthong onset is better controlled than diphthong offset, and more importantly, diphthong production is constrained by the general articulatory-to-acoustic relations. It is generally concluded that in Ningbo Chinese, rising diphthongs have two targets and can thus be understood as vowel sequences while falling diphthongs have only one dynamic target and should be treated as a single vowel phoneme.

**3:45**

**4pSCa7. Regional effects on Indian English sound and timing patterns.** Hema Sirsa (Linguist., Univ. of Oregon, 179 NW 207th Ave., Beaverton, OR 97006, hsirsa@uoregon.edu)

English, spoken as second/third language by millions of speakers of India (IE), differs from other varieties of English in terms of sound patterns. Most descriptions of IE have focused on the influence of native language on IE (Wiltshire and Harnsberger, 2006; Sirsa and Redford, submitted). Some studies have also pointed out that IE may be evolving into multiple varieties due to social and political pressures (Wiltshire, 2005), but so far dialectal differences have not been explored independently from L1 influences. The current study aimed to do just this. Regionally based segmental and suprasegmental differences were investigated in IE spoken by Hindi and Telugu speakers, with equal numbers of speakers of each L1 recruited from two geographical sites (Delhi and Hyderabad). Analysis of IE sound patterns indicated that speakers from Hyderabad had more fronted /u/ than Delhi speakers, whereas Delhi speakers had longer phrase-final lengthening then Hyderabad speakers. Speakers from the two sites also had different rhythm structures and speech rates. These results support the suggestion that IE is evolving into multiple varieties, and that these varieties are not simply a function of different L1s.

**4:00**

**4pSCa8. Tonal alignment in Deori.** Shakuntala Mahanta (Dept. of Humanities and Social Sci., Indian Inst. of Technol. Guwahati, Guwahati, Assam 781039, India, shakunmahanta@gmail.com), Indranil Dutta (Dept. of Computational Linguist., English and Foreign Lang. Univ., Hyderabad, Andhra Pradesh, India), and Prarthana Acharyya (Dept. of Humanities and Social Sci., Indian Inst. of Technol. Guwahati, Guwahati, Assam, India)

This paper reports on the results from an experiment on tone in Deori, a language spoken by about 20,000 people in Assam (India). Data from 10 speakers where the target word bearing the tonal contrast appeared in the sentence medial position is presented. Time-normalized pitch of different words shows that words may have a lexically specified high or low tone. A high tone may contrast with a low tone, but its phonetic implementation of rise or fall in a disyllabic word depends on whether the syllable on which the contrast appears is initial or final. A tonal contrast on the first syllable leads to a falling contour, but when the contrastive tone appears on the second syllable of a disyllabic word then the tonal contour is falling. Exceptions to this pattern appear in closed disyllables where a steep rise in either the low or high tone is not observed. A high or a low tone may also contrast with a word which is not specified with any tone, in which case there is no rise or fall. Statistical analyses show that Deuri tones exhibit phonetic properties that are dependent on contextual factors like syllable position and segmental properties.

**4:15**

**4pSCa9. An acoustic description of Chemehuevi.** Benjamin V. Tucker (Linguist., Univ. of AB, 4-32 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, bvtucker@ualberta.ca)

Chemehuevi, a member of the Uto-Aztecan language family, is spoken along the Colorado River in both Arizona and California. The language is extremely endangered with fewer than five known speakers, all over the age of 50. Chemehuevi is classified following Miller *et al.* (2005) as a dialect of Colorado River Numic along with Southern Paiute and Ute. The present work offers a general description of the acoustic characteristics of the Chemehuevi phoneme inventory based on both an analysis of archival (3 female speakers recorded by: Major, 1969; Tyler, 1972; Press, 1973–1974) and current field recordings (1 male speaker recorded by: Penfield, Serratos,

and Tucker, 2005–2006, 2010) of the language. To date, there is little acoustic analysis of Numic languages available. Vowel characteristics are analyzed by extracting duration and the first three formant frequencies. Consonants are also investigated using relevant acoustic measures (such as voice-onset time and centroid frequency). Additionally, the present acoustic analysis is compared to early descriptions of the phoneme inventory and provides evidence regarding the nature of the vowel inventory (is /e/ a phoneme), location of stress, idiolectal differences, and word final voiceless vowels.

**4:30**

**4pSCa10. Acoustic features of upper necaxa totonac ejective fricatives.** Rebekka Puderbaugh and Benjamin V. Tucker (Dept. of Linguist., Univ. of AB, 2-40 Assiniboia Hall, University of AB, Edmonton, AB T6G 2E7, Canada, puderbau@ualberta.ca)

The purpose of this study is to investigate the acoustic properties of a class of sounds known as ejective fricatives in Upper Necaxa Totonac (UNT), a Totonac-Tepehua language of northern Puebla, Mexico, and to relate these sounds to those in other languages. Ejective fricatives are an exceedingly rare class of sounds found in only a relatively small number of the world's languages. This study attempts to clarify the nature of the acoustic signal of these sounds in UNT, whose historical origins have been reconstructed as former fricative plus glottal stop clusters [Beck, 2006, Univ. of Alberta Working Papers, 1], use the acoustic data to verify whether these segments are in fact canonical ejectives and propose future directions for further research. Analyses of the segments in question include duration and center of gravity of the fricative portions, presence or absence of any periods of silence surrounding the segments, durations of such silences, and effects on pitch, amplitude, duration, and formants of neighboring vowels. Due to the variable nature of the realization of laryngeal phonemes in UNT, pitch, amplitude, and voice quality of both preceding and following vowels were analyzed as well.

**4:45–5:00 General Discussion**

THURSDAY AFTERNOON, 5 DECEMBER 2013 PLAZA A, 1:00 P.M. TO 5:00 P.M.

## Session 4pSCb

## Speech Communication: Speech Production II (Poster Session)

Jelena Krivokapic, Chair
*Linguist., Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109-1220*

All posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors the opportunity to view other posters authors of odd-numbered papers will be at their posters from 1:00 p.m. to 3:00 p.m. and authors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

### *Contributed Papers*

**4pSCb1. Phonological encoding and articulatory duration in spontaneous speech.** Melinda Fricke (Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720-2650, melindafricke@berkeley.edu)

Many studies have found that word duration is correlated with a word's contextual predictability in conversational speech. Lindblom (1990)'s Hypo/Hyperarticulation Theory, Jurafsky *et al.* (2001)'s Probabilistic Reduction Hypothesis, and Aylett and Turk (2006)'s Smooth Signal Redundancy Hypothesis all suggest that such differences in duration are due to processes occurring primarily at the lexical level. The present study, however, suggests that these differences may be attributable to processes occurring at the phonological level. In this study, mixed modeling is used to examine the voice onset time and rime duration of monosyllabic /p t k/ words in spontaneous, connected speech (the Buckeye Corpus; Pitt *et al.*, 2007). Higher contextual predictability given the previous word is found to be associated with shorter VOT, while higher contextual predictability given the following word is associated with shorter rime duration. VOT also varies according to the number and type of a word's phonological neighbors; words with more neighbors overlapping in the rime have significantly longer VOT, while words with more neighbors overlapping in the initial CV have significantly shorter VOT. These results motivate a model of speech production that assumes both lexical-phonological feedback and positional encoding of segments (e.g., Sevald and Dell, 1994).

**4pSCb2. Syntactic probability affects morpheme durations.** Clara Cohen (Linguist., Univ. of California at Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, cpccohen@berkeley.edu)

This project investigates the role of syntactic predictability on the duration of morphemes. Previous research has found that contextually predictable speech units tend to be shorter in duration. Usually, such research focuses on the duration of words or syllables, and context is defined in terms of n-gram strings. This project extends such research by investigating the role of syntactic context on the production of morphemes. Are more probable morphemes also reduced when they are more probable in a given syntactic context? Russian sentences with quantified subject noun phrases (e.g., "three chairs") allow both singular and plural verb agreement suffixes, but the probability of observing one or the other is variable. In this study, Russian speakers produced sentences with either singular or plural agreement of varying probability. The lists were counterbalanced so that each sentence was produced with both the singular and plural suffix. Although there was no difference in duration for singular suffixes, high-probability plural suffixes were shorter than low-probability plural suffixes. Differences in whole-word durations cannot account for these differences in suffix durations. These results suggest that contextual predictability in the form of agreement relations can affect the phonetic production of the morphemes that encode those relations.

**4pSCb3. Reduction and frequency analyses of vowels and consonants in the Buckeye speech corpus.** Byunggon Yang (English Education, Pusan National Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

In a casual conversation American speakers tend to talk fast and to reduce or change sounds of phonetic symbols defined in an English dictionary which we would find in citation speech style. This study examined how much reduction of pronunciation Americans make from the dictionary prescribed symbols to the real speech ones and how frequently Americans use vowels and consonants in the Buckeye speech corpus. The corpus was recorded by 40 American male and female subjects for an hour per each subject. Results were as follows: First, the Americans produced a reduced number of vowels and consonants in daily conversation. The reduction rate from the dictionary transcriptions to the real transcriptions was around 38.2%. There was not much difference between the vowels and consonants in the reduction. Second, the Americans used more front high and back low vowels while 78.7% of the consonants accounted for stops, fricatives, and nasals. This indicates that the segmental inventory has nonlinear distribution in the speech corpus. From those results we conclude that there is a substantial reduction in the real speech from the dictionary symbols and suggest that English educators consider pronunciation education reflecting the real speech data.

**4pSCb4. The effect of high and low variability conditions on phonetic convergence.** Grant McGuire (Linguist., Univ. of California, Santa Cruz, Stevenson Faculty Services, Santa Cruz, CA, gmcguir1@ucsc.edu), Molly E. Babel, and Jamie Russell (Linguist., Univ. of Br. Columbia, Vancouver, BC, Canada)

Studies of phonetic convergence using single-word auditory naming tasks offer insight into how variability in stimuli affect the translation from speech perception to speech production. In this paper, we report on an experiment which compares phonetic convergence in single-word production between high variability (mixed talker condition) or low variability (blocked talker condition) using five female model talkers' voices for the task. Twenty female participants participated in a production task where they produced baseline tokens and shadowed model talker productions in either the high or low variability condition. Phonetic imitation was quantified using listener judgments in an AXB similarity rating task where a model token was compared to a shadower's baseline and shadowed token. The results indicate a trend towards more convergence in the low variability condition, but this was highly affected by model voice; one model voice was spontaneously imitated more in the high variability condition than the low variability condition. Several socio-cognitive tests were administered to shadowers, and continued analyses of the data will explore whether these individual socio-cognitive measures predict shadowers' predispositions toward phonetic convergence.

**4pSCb5. The effect of task difficulty on phonetic convergence.** Jennifer Abel (Linguist., Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, jennifer.abel@alumni.ubc.ca)

Cognitive workload is the information processing load a person experiences when performing a task; the more difficult the task, the greater the cognitive workload. Increased task difficulty/cognitive workload has been shown to have an effect on several acoustic measures of speech such as amplitude, word/syllable/utterance duration, and f0. To date, the task difficulty-speech production link has only been studied in individuals. This study examines the effect of different levels of task difficulty on phonetic convergence within dyads collaborating on a task. Dyad members had to build identical LEGO® constructions without being able to see each other's construction, and with each member having half of the picture-based instructions required to complete the construction. Three levels of task difficulty were created, based on the number of pieces in each step of the construction—easy (2 pieces/step), medium (3 pieces/step), and hard (4 pieces/step)—with five dyads at each level (30 participants total). Dyads were audio- and video-recorded, and completed working memory and mental rotation tests and personality questionnaires prior to the task. Acoustic analysis and AXB perception studies are underway to examine the amount and type of convergence in each dyad.

**4pSCb6. Word-internal ambisyllabic consonants are codas.** Karthik Durvasula, Ho-Hsin Huang, and Rose Merrill (Michigan State Univ., B330 Wells Hall, East Lansing, MI 48824, durvasul@msu.edu)

The syllabic affiliation of ambisyllabic consonants (e.g., the word-medial consonants in happy and Danny) is unclear. Research on ambisyllabic consonants has revealed an inconsistent set of phonetic correlates (Krakow, 1989; Turk, 1994; Gick, 2004). While some suggest they behave as onsets or codas (but not both simultaneously), others suggest their gestural durations are intermediate between onsets/codas. At least some of the research is based on comparisons of the ambisyllabic consonants to word-edge onsets/codas. However, comparisons to word-edges are confounded by the fact that such consonants undergo domain-edge related changes (Fougeron, 2001; Keating *et al.*, 2003b). Here, we control for this confound, and compare ambisyllabic consonants to word-medial onsets and codas. We conducted an experiment on 10 native English speakers, who produced 15 repetitions at three different speech rates of 16 English words (8 test, 8 filler) that consisted of the nasal consonants [n or m] in one of four positions: word-medial onset, word-medial coda, word-final coda, and ambisyllabic context (e.g., gamete, gamble, gam, and gamma). The results suggest: (1) Consistent with previous research, there are durational differences between word-medial and word-final nasal codas; (2) Ambisyllabic consonants clearly pattern with the word-medial nasal codas and are significantly different from the nasal onsets.

**4pSCb7. The articulation of derived affrication in American English.** Jae-Hyun Sung (Linguist,, Univ. of Arizona, 814 E 9th St., Apt. 14, Tucson, AZ 85719, jhsung@email.arizona.edu)

Affrication of coronal stops before /ɹ/ is commonly observed in English. For instance, /t/ in "tree" and /d/ in "dream", in which coronal stops precede /ɹ/, are often realized as affricated stops (i.e., [ʧɹi] instead of [tɹi]; [ʤɹim] instead of [dɹim]). Given that morphological structures and frequency of words play a critical role in many coarticulatory processes (Bush, 2001; Ernestus *et al.,* 2006; Myers and Li, 2009), the present study investigates whether the degree of derived affrication before /ɹ/ is influenced by different morphological structures and frequency of words and phrases. This study uses ultrasound imaging and audio recordings of seven native speakers of American English to examine the articulatory aspect of derived affrication. Comparisons of the degree of affrication show significant differences among words in various environments, in which tautomorphemic words and high-frequency words and phrases lead to greater degree of affrication. Furthermore, the gestural patterns of various morphological and frequency conditions are highly individualized.

**4pSCb8. Temporal coordination of sibilants in Polish onset clusters.** Manfred Pastätter and Marianne Pouplier (Inst. of Phonet. and Speech Processing, LMU, Schellingstraße 3, Munich 80799, Germany, manfred@phonetik.uni-muenchen.de)

In this study we employ the gestural syllable model to examine cluster-vowel timing in Polish sibilant initial (SI = {/ʃm-, ʃp-, sp-, sk-/}) and sibilant final (SF = {/mʃ-, pʃ-, ps-, ks-/}) onset clusters. In this model, the timing of a complex onset is evaluated relative to a simplex onset and it is predicted that timing relations between onset and vowel should be invariant independently of onset complexity (Browman and Goldstein, 2000). Articulatory data of three speakers show that SI clusters conform to the predicted timing pattern in terms of a globally organized onset cluster relative to the vowel ("c-center"). This is compatible with previous findings for several languages. For SF clusters, however, there are considerable timing differences between complex and corresponding simplex onsets. This suggests that SF clusters are coordinated differently (and inconsistently) to the following vowel compared to SI clusters, as also reported previously for Romanian (Marin, 2013). We investigate to which extent this difference between SI and SF clusters is related to sibilants' high coarticulatory resistance preventing a close C-V coordination. We will present an analysis of jaw movement data to consider possible effects of jaw position constraints on the temporal coordination of clusters.

4p THU. PM

**4pSCb9. Compensatory vowel shortening before complex coda clusters in the production and perception of German monosyllables.** Sandra Peters and Felicitas Kleber (Inst. of Phonet. and Speech Processing, LMU, Schellingstr. 3, Munich 80799, Germany, sandra@phonetik.uni-muenchen.de)

The main aim of the present study was to investigate incremental coda compensatory shortening in the production and perception of German monosyllables including factors such as accentuation (i.e., accented vs deaccented) and codas' manner of articulation (i.e., sonorant vs obstruent). Ten speakers produced real German words like /klɪŋ/ and /klɪŋt/. We measured the duration of the vowel and the first coda consonant (C1). Overall there was no significant vowel shortening effect. However, some speakers did show vowel shortening and even more so in accented tokens with sonorant codas. Additionally, all speakers tended to shorten C1. In a subsequent experiment, we tested whether listeners compensate for different degrees of vowel and C1 shortening. 21 subjects judged which vowel in selected pairs such as /klɪŋ/—/klɪŋt/ was longer. In two thirds of all pairs, listeners perceived vowels before simplex codas as longer—even in pairs with equal segment durations. While this overall bias indicates perceptual vowel shortening before complex codas, listeners nevertheless show tendencies to compensate for non-shortened vowels before complex sonorant codas, i.e., they were perceived as longer. Although there was less vowel shortening in production, listeners showed perceptual vowel shortening and some tendencies toward compensation in contexts that favor shortening.

**4pSCb10. Revisiting the consonantal voicing effect: Flapping in American English.** Ylana Beller-Marino and Dianne Bradley (Linguist., CUNY Graduate Ctr., 360 1st Ave., Apt. 6D, New York, NY 10010, ybeller@gc.cuny.edu)

It is long-acknowledged that the consonantal voicing effect—whereby vowel duration is greater preceding voiced vs voiceless consonants (e.g., *rib*/*rip*)—is larger in English as compared with other languages, and that the effect's magnitude generally decreases in multisyllabic forms (e.g., *rabbit*/*rapid*). The current study examines consonantal voicing effects in the multisyllabic environment, crucially contrasting non-coronal with coronal cases (e.g., *rider*/*writer*). In American English, the latter are subject to a flapping process that surface-neutralizes the voicing distinction. Hence, while both phonological and phonetic sources for a vowel-duration difference are available in non-coronals, flapping eliminates the phonetic source in coronals. We present an analysis of critical vowel durations in elicited productions (target words uttered in a carrier phrase), and confirm the pattern expected if the post-vocalic consonant's place of articulation matters: the consonantal voicing effect was entirely reliable for non-coronals, but not for coronals. More detailed analyses set aside tokens where flapping failed to apply, and found that the consonantal voicing effect might be altogether absent with coronal place. We speculate that, here, the voicing distinction may have been neutralized in phonological representation, whether that distinction is a matter of orthography (*doodle*/*duty*) or is also supported by morphological alternation (*rider*/*writer*).

**4pSCb11. Phonetics as a complement to phonology in the Canadian Shift.** Matt H. Gardner (Linguist., Univ. of Toronto, Toronto, ON, Canada) and Rebecca Roeder (English, Univ. of North Carolina at Charlotte, 9201 University City Blvd., Charlotte, NC 28223, rroeder@uncc.edu)

Previous accounts of the Canadian Shift have interpreted this diachronic change in vowel pronunciation as a purely phonetic consequence of the low back LOT-THOUGHT vowel merger; however, such an analysis does not transparently explain the strong connection between the (phonological) low back merger and the subsequent (phonetic) retraction of the TRAP vowel in the acoustic vowel space. This paper addresses this issue by presenting an analysis of the shift that combines the approaches of Modified Contrastive Specification theory and the Contrastive Hierarchy—two phonological frameworks—with phonetic insights from Vowel Dispersion-Focalization theory. We propose that the catalyst of the Canadian Shift is a three-way vowel merger, in combination with a simultaneous change in the underlying feature specifications of the TRAP vowel. This results in a phonology that allows for the TRAP and DRESS vowels to succumb to the influence of the phonetic principles of dispersion and focalization. This hypothesis is illustrated by comparison of data from 59 speakers in Thunder Bay, Ontario, and Industrial Cape Breton, Nova Scotia. Our analysis predicts that a Canadian Shift-type phonetic change will occur in any North American dialect of English where the PALM-LOT-THOUGHT merger occurs, unless an intervening phonological change alters systemic contrasts.

**4pSCb12. Perceptual and prosodic factors in cluster timing: Manner, order, and syllable position effects in Polish consonant clusters.** Marianne Pouplier and Manfred Pastätter (Inst. of Phonet., LMU, Schellingstr. 3, Munich 80799, Germany, pouplier@phonetik.uni-muenchen.de)

We investigate timing in Polish tautosyllabic C1C2 clusters differing in manner, consonant order, and syllable position. Hoole *et al.* (2013) reported for German that perceptual constraints may condition timing differences in /kn/ and /kl/ clusters due to the nasal but not the lateral obscuring the preceding stop burst. Using articulography, we test this hypothesis for a variety of Polish onset clusters (C1={m, p, k}, C2={n, l, r}). Results from three speakers confirm a significant influence of both C1 and C2 on timing patterns. A C1 nasal shows more overlap than a stop. For C2, /l/ shows more overlap than /n/, consistent with the German results. However, the relative difference between C2=/n/ and C2=/l/ holds independently of whether C1 is a nasal or a stop, contra the perception hypothesis. Further, it is known from several languages that onset clusters overlap less than coda clusters, yet this observation has been confounded by the sonority conditioned change in consonant order in onset/coda. Polish has several clusters which do not change order as a function of syllable position, allowing us to tease these two factors apart. If consonant order is kept constant, there is no significant syllable position effect on C-C timing.

**4pSCb13. Entrainment by vocal effort: Coordination in postural control and speech production.** Robert Fuhrman (Dept. of Linguist, Univ. of Br. Columbia, Totem Field Studios 2613 West Mall, Vancouver, BC V6T1Z4, Canada, robert.a.fuhrman@gmail.com), Adriano Vilela Barbosa (Elec. Eng., Universidade Federal de Minas Gerais, Belo Horizonte, Minas Gerais, Brazil), and Eric Vatikiotis-Bateson (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

The biomechanical coupling between the systems implicated in speech production, postural control, and respiration suggests that some degree of coordination in the form of postural entrainment should take place given excessive task demands in the speech domain, as has been previously reported [Vatikiotis-Bateson, *et al.*, (2009) Proceedings of ESCOM 2009]. In this context, this work assesses the time-varying coordination and entrainment among multiple components of the postural control system that result from the modulation of vocal effort level in both read and spontaneous speech. Correlation map analysis is used to quantify the coordinated patterns of interaction between speech acoustics and a variety of related physical systems, including lower body postural configuration (center-of-pressure calculated from force plate measurements), head motion (measured with OPTOTRAK), and visual motion (optical flow from video). Cross-correlation analysis of the measured signals shows that modulation of vocal effort leads to both qualitative and quantitative shifts in the coordinative dynamics of the system, uniformly resulting in better spatiotemporal coordination and reduced rhythmic pattern complexity as vocal effort is increased.

**4pSCb14. Acoustic correlates of consonant gesture timing in English.** Elliot Selkirk and Karthik Durvasula (Linguist. and Lang., Michigan State Univ., B331 Wells Hall, East Lansing, MI 48824, selkirke@msu.edu)

There is extensive research on the organization of syllable-structure as indexed by the relative timing of the articulators (Browman and Goldstein, 1988; Byrd, 1995; Shaw *et al.*, 2011 inter alia). The research suggests consonants in complex onsets (in words such as scream, stream…) are aligned to a single position called the C-center, the mean of the midpoints of the onset consonants. However, such research typically uses very expensive articulatory equipment (X-ray Microbeam, Electromagnetic Articulography…). This restricts the research to a few laboratories across the world with access to such technology. Here, we explore the possibility of using acoustic measurements, which are cheaper and more accessible, for

such research. We conducted an experiment on 6 native speakers of English, who produced 12 repetitions of 24 English words (12 test, 12 filler) that varied in the number of onset consonants (C1, C1C2, C1C2C3) in three different vowel contexts. Paralleling previous studies, the results show that onset consonants align with the C-center even in acoustic measurements. The results suggest acoustic data has at least some meaningful information about gestural organization. Therefore, they prompt the (nuanced) use of acoustic techniques to study such effects.

**4pSCb15. Coarticulation and contrast in static and dynamic models of second formant trajectories.** Indranil Dutta (Dept. of Computational Linguist., The English and Foreign Lang. Univ., Tarnaka, Osmania University Campus, Hyderabad 500605, India, indranil.dutta.id@gmail.com) and Charlie Redmon (School of Lang. Sci., The English and Foreign Lang. Univ., Hyderabad, Andhra Pradesh, India)

Real (Stevens *et al.*, 1966) and virtual F2 locus (Sussman *et al.*, 1991) measures are presented for Malayalam lingual plosives. We show that in distinguishing voiceless coronals (dental, alveolar, and retroflex) in VC:V sequences, F2 onsets derived from first-order locus equations (LEs) show only partial delineation of the contrast. The dental-alveolar contrast is effectively maintained, but retroflex and alveolar stops show no significant difference in F2 onset. Following Lindblom and Sussman's (2012) examination of LEs as a measure of relative coarticulatory resistance, we report F2 slopes for the three coronal stops in VC and CV transitions to assess the implications of this metric in Malayalam. Our findings on the ordering of slope values from steepest to flattest did not follow predictions based on expectations of relative articulatory complexity; namely, alveolars generated a flatter slope than retroflexes, despite Dart and Nihalani's (1999) demonstration that the retroflex gesture is more complex within the coronals. These results, when compared with temporal measures from exponential models of formant trajectories at consonant implosion and release (i.e., transition velocity and projected F2 locus), suggest a necessary distinction between coarticulation-based place of articulation categorization and formant transition cues utilized in maintaining stop place contrasts.

**4pSCb16. Estimation of vocal tract input impedance at the glottis from formant measurements.** Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 4789 N White River Dr., Bloomington, IN 47404, slulich@indiana.edu)

It is well known that the mapping from articulation to acoustics is many-to-one or many-to-many, so that so-called "articulatory-acoustic inversion" is a challenging problem. What has not been noted, however, is that the input impedance from the glottis can be determined from the inverted articulatory configuration regardless of whether this configuration is accurate. This can be useful as a step toward estimating the acoustic load on vocal fold vibration during phonation. The theory and procedure for thus obtaining estimates of the vocal tract input impedance is presented, its relation to the Mermelstein/Schroeder method is shown, and its limitations are discussed. Finally, experiments with synthetic and naturally produced vowels are presented and discussed. It is shown that the estimated input impedance is accurate up to the highest measured formant, with the largest deviations centering around the formant frequencies due to errors in formant measurements and the handling of acoustic losses.

**4pSCb17. Modeling the listener? What resets acoustic durations of repeated English words.** Prakaiwan Vajrabhaya and Vsevolod Kapatsinski (Linguist., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, pvajrabh@uoregon.edu)

Listener-based accounts of speech production claim that speakers modify their speech based on their evaluation of the listener's state of knowledge (Lindblom, 1990). In line with this, repeated words shorten when they have been previously said to the same listener (Fowler, 1988); however, repetition across an episode boundary in a narrative does not lead to decreased acoustic duration (Fowler *et al.*, 1997). We replicate Fowler *et al.*'s story boundary effect and extend the study by testing whether a switch in listener has an additional effect on word duration. Speakers were asked to tell and retell the same story in the sequence of (A) listener 1/(B) listener2/(C) listener1 again (Galati and Brennan, 2010). We expect word durations to reset when the speaker starts a new narrative, especially when there is a switch in listener. In other words, word durations should be comparable in conditions (A) and (B), but shorter in (C), since the listener in condition (C) has heard the story before. Acoustic data from 20 American English native speakers have been collected and transcribed; data analysis is ongoing. This study is intended to shed light on the interplay between production economy and the need to transmit information.

**4pSCb18. Acoustic analysis of initial consonants in the California Syllable Test.** E. W. Yund, Marc Ettlinger (Res. 151/MTZ, VA Medical Ctr., 150 Muir Rd., Martinez, CA 94553, yund@ebire.org), and David L. Woods (Neurology, VA Medical Ctr., Martinez, CA)

The goal of the present study is to conduct an acoustic analysis of onset consonants to identify the spectrotemporal features that distinguish them from each other and to identify acoustic consonant variations that produce the observed patterns of perceptual confusions seen in young- and older-normal-hearing listeners [J. Acoust. Soc. Am. **127**, 1609–1623 (2010); JRRD **49**, 1277–1292 (2012)]. We used the California Syllable Test (CaST) token set, which includes 40 exemplars of each initial consonant for each of three vowels and six talkers. The CaST measures recognition of 20 initial and 20 final consonants in speech-spectrum noise with each consonant presented at a 67%-correct signal-to-noise ratio (SNR). Time-normalized spectrograms are computed for each exemplar (from consonant onset to the end of the formant transition) by varying the time-spacing of the FFT spectral lines in proportion to the exemplar duration. Quantitative comparisons among normalized spectrograms of correctly recognized and confused exemplars at a range of SNRs suggest explanations for specific consonant confusions. The long-term goal is to apply this analysis to understand the effects of hearing loss and HAs on consonant perception and to predict consonant confusion patterns obtained with the CaST in normal-hearing and hearing-impaired listeners.

**4pSCb19. The role of the posterior cricoarytenoid muscle in phonation.** David Berry, Dinesh K. Chhetri, and Juergen Neubauer (Surgery, UCLA, 31-24 Rehab., Los Angeles, CA 90095-1794, daberry@ucla.edu)

The posterior cricoarytenoid muscle (PCA) is generally considered to be a respiratory muscle. Indeed, as the sole abductor of the glottis (i.e., the only laryngeal muscle with the capability of opening the true vocal folds), paralysis of the PCA may lead to asphyxiation. While the PCA muscle also appears to play a role in phonation, a consensus has not been reached among voice scientists regarding its precise role in the control of fundamental frequency, phonation threshold pressure, and other phonatory variables. Using a new developed method of graded stimulation to the laryngeal muscles, Chhetri, Neubauer, and Berry (2012) explored the role of the cricothyroid muscle (CT), thyroarytenoid muscle (TA), and the lateral cricoarytenoid and interarytenoid muscle complex (LCA + IA) on fundamental frequency, phonation threshold pressure and glottal posturing. The present study augments the previous study by also investigating the influence of the PCA muscle on these same phonatory variables. Similar to the adductor muscles, it is shown that the PCA muscle introduces new possibilities for achieving multiple phonation types at a given fundamental frequency.

**4pSCb20. Anatomic development of the hyo-laryngeal complex in humans from birth to 95 Years: An imaging study.** Houri K. Vorperian (Waisman Ctr., Univ. of Wisconsin, Waisman Ctr., 1500 Highland Ave., # 427, Madison, WI 53705, vorperian@waisman.wisc.edu), Yuan Wang (Dept. of Statistics, Biostatistics & Medical Informatics, Univ. of Wisconsin, Madison, WI), Reid B. Durtschi (Waisman Ctr., Univ. of Wisconsin, Madison, WI), Meghan M. Cotter (Dept. of Neurosci., Univ. of Wisconsin, Madison, WI), Ray D. Kent (Waisman Ctr., Univ. of Wisconsin, Madison, WI), Moo K. Chung (Dept. of Statistics, Biostatistics & Medical Informatics, Univ. of Wisconsin, Madison, WI), and Lindell R. Gentry (Dept. of Radiology, Univ. of Wisconsin, Madison, WI)

During postnatal development, the hyo-laryngeal complex descends in the pharyngeal cavity primarily during early childhood, followed by a secondary descent during puberty, particularly in males. The purpose of this

study is to quantify the descent of the human hyo-laryngeal complex, as well as its relational growth to other functionally related structures such as the epiglottis and the tongue from birth to 95 years. Anatomic data secured from 902 medical imaging studies (482 males; 420 females) were analyzed in two phases: (I) A detailed assessment of developmental changes of the hyo-laryngeal complex and functionally related structures from birth to 19 years using 771 imaging studies. I(I) Comparison of similar measurements between three adult groups (ages 20-to-45 years; 45–70 years, and 70–95 years) using 131 images. Findings indicate that: (a) growth/descent of the hyo-laryngeal complex is non-linear and protracted, displaying a predominantly somatic growth pattern; (b) small sex differences in growth are present during childhood, with increased differences emerging at about 10 years, and maximal differences present by 19 years; and (c) there appears to be a coincident relational growth of functionally related structures. These novel findings are of clinical significance, and enhance the understanding of vocal tract development. [NIH-Grants R01DC6282, P-30HD03352.]

**4pSCb21. Signal detection of lipreading visemes using two dimensional and three dimensional images.** Rita Quigley and Al Yonovitz (The Univ. of Montana, 32 Campus Dr., Missoula, MT 59812, rita.quigley@mso.umt.edu)

The actual process by which the lipreader translates the lip movements they identify into a message is very complex. The lip movements observed represent only fragments of the complete message. The main purpose of this study is to investigate (1) the ability of lipreaders to use visual information alone to identify phonemes in varying contexts including nearby coarticulation effects and vowel neighborhoods; (2) lipreading responses using the effect of improved video presentation through 3D video, providing better and more realistic video presentation; and (3) the use of a novel measurement technique, i.e., a signal detection two-alternative-forced choice method of subject response that should provide measures of discrimination between phonemes including "visemes." Video recordings were made in both 2D and 3D formats. This 3D image presented more realistically the movements such as lip-rounding and micro-movements of viewable articulators in three dimensions. Subjects with normal hearing were presented these video presentations. A Two-Alternative-Forced-Choice (2AFC) paradigm was used. The consonants were viewed with various vowel contexts. D-prime values were obtained for both the 2D and 3D videos. Particular consonant clusters were more discriminable in 3D.

**4pSCb22. Speaking tongues are always braced.** Bryan Gick, Blake Allen (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T1Z4, Canada, gick@mail.ubc.ca), Ian Stavness (Comput. Sci., Univ. of SK, Saskatoon, SK, Canada), and Ian Wilson (CLR Phonet. Lab., Univ. of Aizu, Aizuwakamatsu, Japan)

Bracing the tongue against rigid vocal tract surfaces (i.e., teeth or palate) has been suggested to be important in facilitating certain kinds of tongue movements [Stone, J. Acoust. Soc. Am. **81**, 2207–2218 (1990)]. However, previous studies have generally sought bracing in only a narrow range of phonetic contexts, resulting in a widespread view of bracing as an occasional state, peculiar to specific sounds or sound combinations. The present study uses electropalatography (EPG) as well as ultrasound imaging and electromagnetic articulometry (EMA) to describe tongue bracing in continuous speech passages, finding that the tongue is almost constantly braced against lateral surfaces during running speech. Analysis of archival data from the male and female speakers of American English in the KayPENTAX Palatometer Database (Model 4333) shows that they brace the tongue continuously, except during a small percentage of low vowels, and during a larger percentage of instances of /l/. Additional measures using all three devices, as well as biomechanical simulations using ArtiSynth (www.artisynth.org), provide further insight, indicating that the tongue also braces against the central palate and/or lower jaw, and that bracing points slide anteroposteriorly across speech sounds. These results suggest that bracing is a constant and necessary aspect of tongue motor control.

**4pSCb23. Direct characterization of collagen recruitment in the human vocal fold lamina propria.** Bahar Fata (Head & Neck Surgery, UCLA, 1000 Veteran Ave., Rm. 33-59, Los Angeles, CA 90024, bahar.fata@gmail.com), Julio L. Vergara (Physiol., UCLA, Los Angeles, CA), and Zhaoyan Zhang (Head & Neck Surgery, UCLA, Los Angeles, CA)

The goal of this study is to develop a structurally based constitutive model to characterize the structure-function relationship of the vocal folds. Compared to phenomenological models, structurally based constitutive models allow direct prediction of changes in the mechanical behavior of the vocal folds as a result of aging or pathological conditions. The first significant step in developing such a mathematical model is to characterize the structural arrangement and load-bearing behavior of the collagen and elastin fibers, the two most mechanically significant structural proteins in the vocal fold. A micro horizontal uniaxial tensile system has been designed and coupled with the non-invasive multi-photon microscopy method. The load-bearing or recruitment behavior of collagen was characterized by simultaneously measuring the waviness of the collagen fibers and stress of the cover layer at different strain conditions. The structural arrangement of the collagen and elastin fibers in the different layers of the lamina propria were also quantified. The results of this study will directly elucidate the specific contributions of the elastin and collagen fibers to the vocal fold mechanical behavior under uniaxial tension. [Work supported by NIH.]

**4pSCb24. Electropalatography examination of groove width in Russian.** Phil Howson (Linguist., The Univ. of Toronto, 100 St. George St., Toronto, ON M5S 3G3, Canada, phil.howson@mail.utoronto.ca)

Previous studies have indicated a difference between voiced and voiceless pairs of consonants with respect pre-constriction vocal tract volume. This article utilizes electropalatography (EPG) to examine the anterior and posterior groove width of palatalized and non-palatalized fricative pairs in Russians in order to observe different degrees of pre-constriction vocal tract volume. Measurements were taken at the point of maximum constriction using Articulate Assistant software. Higher degrees of contact with the palate were taken to indicate smaller pre-constriction vocal tract volume. The results (based on a single speaker), indicate a significant difference in the degree of contact with the palate between the voiced and voiceless pairs of non-palatalized fricatives. However, the palatalized consonants indicated no significant difference in the degree of contact with the palate. The findings suggest that the smaller vocal cavity created by the secondary articulatory gesture for palatalization is sufficient to facilitate voicing and frication; in the case of the voiced fricatives, the sub-glottal pressure is adjusted to permit vibration of the vocal cords. The findings further suggest that speakers adhere to the principle of minimal articulatory effort when producing speech.

**4pSCb25. An analysis of tongue shape during parkinsonian speech.** Katherine M. Dawson (Speech-Language-Hearing Sci., City Univ. New York Graduate Ctr., 365 5th Ave., New York, NY 10016, kdawson2@gc.cuny.edu), Khalil Iskarous (Linguist., Univ. of Southern California, Los Angeles, CA), and D. H. Whalen (Speech-Language-Hearing Sci., City Univ. New York Graduate Ctr., New Haven, Connecticut)

Parkinson's disease (PD) is a neurological disorder characterized by the degeneration of dopaminergic neurons. Speech impairments in PD are characterized by slowed muscle activation, muscle rigidity, variable rate, and imprecise consonant articulation. Complex muscular synergies are necessary to coordinate tongue motion for linguistic purposes. People with PD may show an altered rate of change in tongue shape during vowel to consonant transitions and may also ultimately attain less complex consonantal tongue shapes than controls during speech. In order to test this hypothesis, five PD participants, five older controls and five younger controls (all French-speaking) were imaged using ultrasound. They produced consonant–vowel–consonant word stimuli. Transitions analyzed were vowel-to-liquid (/l/) and vowel-to-velar stop. Tongue shapes were analyzed using a method designed to infer complexity by analogy with the bending energy of a thin shell [Young, Walker, and Bowie, Info. Control **25**(4), 357–370 (1974)]. This method works by integrating the squared curvature of a piece-wise polynomial function fitted to the extracted discrete tongue contour. Results will be discussed in terms of shape change during the transition and maximal consonantal shape attained between PD and control subjects.

**4pSCb26. Characterizing post-glossectomy speech using real-time magnetic resonance imaging.** Christina Hagedorn (Dept. of Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, chagedor@usc.edu), Adam Lammert (Viterbi School of Eng., Univ. of Southern California, Los Angeles, CA), Yihe Zu, Uttam Sinha (Dept. of Otolaryngol., Head and Neck Surgery, Keck School of Medicine, Univ. of Southern California, Los Angeles, CA), Louis Goldstein (Dept. of Linguist, Univ. of Southern California, Los Angeles, CA), and Shrikanth S. Narayanan (Viterbi School of Eng., Univ. of Southern California, Los Angeles, CA)

We investigate articulatory behavior in post-glossectomy speech using real-time magnetic resonance imaging. Our data reveal that listeners judge speech produced by partial-glossectomy patients as atypical when the surgical procedure affected the oral tongue. Speech produced by patients whose procedure affected the base of tongue, however, was judged as typical. We observe that preservation and compensation mechanisms are exhibited by the patients with atypical speech. They preserve appropriate modulation of F1 using tongue and/or jaw height despite inability to appropriately modulate F2 due to the reduced size and/or mobility of the tongue. Further, durational differences between tense and lax vowels are maintained. The preservation of these features serves as evidence in support of a framework within which individual gestural parameters are independently controlled; when achievement of a particular parameter specification (e.g., constriction location) is compromised, the remaining (e.g., constriction degree, activation duration) are unchanged. Compensatory behavior is exhibited when coronal tongue movement has been impeded and is exemplified by (i) production of labiodental stops in place of target coronal stops and laterals and (ii) forming a velar constriction to produce frication in place of the alveolar frication for /s/.

**4pSCb27. Interspeaker variability in relative tongue size and vowel production.** Adam Lammert (Signal Anal. and Interpretation Lab., Univ. of Southern California, 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089, lammert@usc.edu), Christina Hagedorn (Dept. of Linguist., Univ. of Southern California, Los Angeles, CA), Michael Proctor (Marcs Institute/School of Humanities and Lang., Univ. of Western Sydney, Sydney, NSW, Australia), Louis Goldstein (Dept. of Linguist., Univ. of Southern California, Los Angeles, CA), and Shrikanth Narayanan (Signal Anal. and Interpretation Lab., Univ. of Southern California, Los Angeles, CA)

The tongue varies across speakers in terms of the proportion of the overall speech production apparatus that it occupies. Differences in tongue size have the potential to result in speaker-specific articulatory strategies for shaping the vocal tract area function and, in turn, individual patterns of vowel acoustics. The present study examines the interplay between relative tongue size and vowel production using real-time magnetic resonance imaging with synchronous audio. Two populations of native American English subjects are considered, one containing healthy adult speakers with no relevant pathologies, and another containing speakers who had undergone glossectomy as treatment for tongue cancer. All subjects were imaged in the midsagittal plane while reading phonetically balanced English sentences. The size of the tongue and the speech production apparatus were quantified from an overall average posture, and their ratio was correlated with the shape of the vowel space in terms of acoustics (e.g., formant frequencies), constrictions (i.e., location and degree of minimum constriction), and parameterized vocal tract cross-distance functions. Results indicate that relative tongue size can be used to explain and predict observable interspeaker differences in vowel production.

**4pSCb28. Control of voice intensity.** Karin Sjögren, Emmma Ström (Dept. Logopedics, Phoniatrics and Audiol., Lund Univ., Lund, Sweden), and Anders Lofqvist (Dept. Logopedics, Phoniatrics and Audiol., Lund Univ., 300 George St., New Haven, Connecticut 06511, lofqvist@haskins.yale.edu)

This study examined the control of voice intensity using acoustic and aerodynamic recordings. A total of 34 subjects participated half of them with and half without song training, 21 females and 13 males. The subjects produced the syllable sequence /papapa/ while the acoustic signal, the oral air flow, and the oral air pressure were recorded using the Kay-Pentax Phonatory Aerodynamic System. The oral pressure provided an estimate of the subglottal pressure. A measure of glottal flow resistance was calculated as the ratio between subglottal pressure and oral air flow.Three different voice levels were used, normal, reduced, and increased; the change between the normal level and the two others was required to be 6–10 dB. Overall, an increase in voice intensity was associated with increased subglottal pressure and glottal flow resistance with only a small increase in air flow. A comparison between the subjects with and without song training showed those with training to produce higher intensities, to use higher subglottal pressure, but lower glottal flow resistance. Female voices had lower subglottal pressure and lower flow rates but higher glottal resistance than male voices.

**4pSCb29. Menstrual cycle-dependent plasticity of auditory-vocal integration in vocal pitch production.** Hanjun Liu, Xiaoxia Zhu, and Yang Niu (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., 58 Zhongshan 2nd Rd., Guangzhou, Guangdong 510080, China, lhanjun@mail.sysu.edu.cn)

Considerable evidence suggests that auditory function can be influenced by gonadal steroids (estradiol and progesterone), but whether there is a sex hormonal modulation of auditory-vocal integration in vocal production remains unknown. The present event-related potential (ERP) study sought to examine the behavioral and neurophysiological processing of auditory feedback during self-produced vocalization across the menstrual cycle. Eleven young Mandarin-native female speakers with regular menstrual cycle were tested during the menstrual, follicular, and luteal phases. Subjects heard their voice pitch-shifted 50 or 200 cents while producing a vowel sound /u/. Vocal compensations and ERPs in response to pitch perturbations as well as estradiol and progesterone concentrations were measured at three different phases. The behavioral findings showed significantly larger magnitude of vocal compensation at the menstrual phase in comparison to follicular or luteal phase. As to the neurophysiological findings, P2 amplitude in the luteal phase was significantly smaller compared to that in the menstrual and follicular phase. These results demonstrate the menstrual cycle-related effect on the behavioral and neurophysiological processing of auditory feedback in vocal pitch production, suggesting that the integration between auditory and vocal motor system can be modulated by the estradiol and progesterone levels across the menstrual cycle.

**4pSCb30. A computer assisted pronunciation training system.** Kwansun Cho and John G. Harris (Elec. and Comput. Eng., Univ. of Florida, University of Florida, Gainesville, FL 32611, kscho@cnel.ufl.edu)

A computer assisted pronunciation training (CAPT) system is implemented for native Korean speakers who are learning American English. The CAPT system is designed to help a Korean adult learner improve his/her production and perception of American English front vowels (/i, I, ε, æ/) since these vowels are the most difficult for Korean learners due to the different phonetic systems of the two languages. The CAPT system provides a learner a learning session mimicking a live interaction between teacher and student as well as a practice session triggering a learner's interest in continued practice. Pedagogically meaningful activities such as listen-and-repeat, minimal-pair-comparison, target-sound-isolation, and record-and-play are utilized in the learning session. During the learning session, the CAPT system analyzes a monosyllabic word including one of the target front vowels spoken by a learner and gives instantaneous personalized feedback. During the practice session, the CAPT system provides real-time games that are fun but also provide the necessary perception and articulation practice.

**4pSCb31. Measurable acoustic variants as predictors of progress in speech therapy.** Kathleen Siren (Speech-Lang. Pathology/Audiol., Loyola Univ. Maryland, 4501 North Charles St., Baltimore, MD 21210, ksiren@loyola.edu)

Despite the availability of free, user-friendly acoustic analysis programs, acoustic documentation of speech sound change during speech therapy is rarely mentioned in speech research literature. Thus, the utility of acoustic analysis to document speech change over time in children with speech errors

is unknown. A prior study documented children's /s/ productions as they progressed through speech therapy and compared spectrographic analysis of productions to clinicians' perceptual judgments of accuracy. Results indicated a greater number of /s/ productions were judged accurate based on visual (acoustic) analysis vs auditory (perceptual) judgment for all clients, particularly during a period of time when clients' /s/ productions were becoming more frequently accurate. The purpose of this current investigation is to identify the measurable acoustic features of /s/ production that indicate when an individual's /s/ production is improving even when the productions are still heard as incorrect. By comparing productions identified as correct visually but incorrect auditorily to productions identified the same both visually and auditorily, this study identifies acoustic variants that are indicative of subtle improvements in production not yet identifiable by adult listeners. These subtle, yet measurable, acoustic characteristics may identify potential acoustic markers for sound maturation in children's disordered speech production.

**4pSCb32. Assessment of head reference placement methods for optical head-movement correction of ultrasound imaging in speech production.** Kevin Roon, Eric Jackson (CUNY Graduate Ctr., 365 Fifth Ave., Ste. 7107, New York, NY 10013, kroon@gc.cuny.edu), Hosung Nam, Mark Tiede (Haskins Labs., New Haven, CT), and Doug H. Whalen (CUNY Graduate Ctr., New York, NY)

One method of quantification of tongue movement using ultrasound imaging during speech production requires determination of tongue position relative to the palate, corrected for probe and head motion so that successive frames can be meaningfully compared. This method involves placing infrared emitting diodes (IREDs) on a "tiara" attached to the participant's head (Whalen *et al*., 2005). An alternative is to attach IREDs directly to the participant's skin. In either case, the IREDs can potentially move relative to the participant's skull. The present study examined movement with both methods for simple utterances, a read paragraph, and spontaneous speech. The amount of IRED movement observed using both methods allowed identification of regions where IREDs should be affixed on a participant's skin to minimize movement when the direct application method is used. Results of simulations showing the effects of this IRED movement on the calculated head-movement correction of the tongue images are presented. Given the results of these simulations, guidelines are proposed for establishing thresholds that can be used to determine whether a given experimental trial should be included based on the amount of reference IRED movement. Differences in movement due to linguistic content or style will also be discussed.

**4pSCb33. Using an exponential sine sweep to measure the vocal tract resonances.** Bertrand Delvaux and David Howard (Dept. of Electronics, Univ. of York, Heslington, York, York YO10 5DD, United Kingdom, bertrand.delvaux@gmail.com)

The vocal tract (VT) of a singer acts as a filter on the acoustic output from the vibrating vocal folds, enhancing several frequency bands whose peaks are called formants. The nature of these formants is characterized by the shape and dimensions of the VT and they are numbered with the first formant being the lowest in frequency. Perceptually, the first (F1) and second (F2) formants indicate the vowel being sung while the third (F3), fourth (F4) and fifth (F5) relate to the timbre or tone color of the output sound. It is therefore relevant to the understanding of the vocal organ to be able to measure the resonances of the tract with precision. Here we apply the exponential sine sweep method used in room acoustics to VT models and replicas. We use an exponential sine sweep as the source signal for the cavity and record its output. After convolving the output signal with the appropriate inverse filter, we can separate the linear impulse response of the tract from its harmonic distortions. This method is both applied on VT models of Chiba and Kajiyama and on MRI-based molded VTs.

**4pSCb34. A comparison of kinematic and acoustic approaches to measuring speech stability between speakers who do and do not stutter.** Eric Jackson (The Graduate Ctr. of the City Univ. of New York, 365 5th Ave., 7th Fl., Rm. 7304, New York, NY 10016, ejackson@gc.cuny.edu), Mark Tiede (Haskins Labs., New Haven, CT), and Douglas H. Whalen (The Graduate Ctr. of the City Univ. of New York, New York, NY)

People who stutter have been found to exhibit reduced speech stability during fluent speech production relative to people who do not stutter. One index for quantifying stability that has been applied to stuttering and nonstuttering speakers is the spatiotemporal index (STI; Smith *et al*., 1995). STI measures the consistency of repeated speech movements aligned using linear normalization. Similar stability indices based on nonlinear methods for alignment have also been reported (e.g., Lucero *et al*., 1997). Both linear and nonlinear methods have been applied to kinematic signals in previous experiments. The present study tests the possibility that measures of stability based on acoustic signals can also be useful indicators of speech stability in adults who do and do not stutter (cf. Howell *et al*., 2009), as using audio recordings to calculate speech variability could provide an attractive alternative for speech-language pathologists and researchers who lack access to kinematic data. In addition, both kinematic and acoustic stability are assessed with respect to effects of linguistic complexity and social factors.

## Session 4pUW

# Underwater Acoustics and Acoustical Oceanography: Sediment Acoustics: Modeling, Measurements, and Inversions I

Nicholas P. Chotiros, Cochair
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**Chair's Introduction—1:15**

## *Invited Papers*

**1:20**

**4pUW1. A discussion of possible measurement techniques for muddy sediments and of the related modeling challenges.** Allan D. Pierce (P. O. Box 339, P. O. Box 339, East Sandwich, MA 02537, adp@bu.edu), Joseph O. Fayton, and William L. Siegmann (Dept. of Mathematics, Rensselaer Polytechnic Inst., Troy, NY)

Present paper draws on recent work of the late William Carey at Dodge Pond, CT, and on related modeling efforts at RPI. Carey affirmed that muddly sediments can have a substantial air bubble content. The water, solid particles (clay), and bubbles lead to a low sound speed that for low frequencies is explained by a modification of the Mallock-Wood formula. Independent measurements of mass density and sound speed should enable estimates of the fractional composition. The attenuation of sound in muddy sediments without bubbles is small, much smaller than of sandy sediments, and this is explained in terms of the card house model because of the very small size of the clay particles. The larger bubbles are randomly dispersed and have flattened shapes (also explained by the card-house model), and lead to scattering and reflection phenomena. Speculations are made as to whether inversion techniques can be devised to determine bubble shapes and size distributions. The small shear modulus of muddy segments has been tentatively explained in terms of electrostatic effects inherent to the card-house model, and this can possibly be measured by interface waves. Paper also suggests that penetrometer measurements, guided by theoretical modeling, may lead to useful inferences.

**1:40**

**4pUW2. The high frequency environmental acoustics sediment model in the light of recent advances.** Nicholas Chotiros and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas, PO Box 8029, Austin, TX 78713-8029, chotiros@arlut.utexas.edu)

The high frequency environmental acoustics sediment model (HFEVA) published in the High-Frequency Ocean Environmental Acoustic Models Handbook (APL-UW 9407), which has been widely adopted by underwater acousticians and sonar modelers, is examined in the light of recent sediment acoustic models and measurements, particularly the multiple scattering and poro-elastic models. The former indicates that the sound speeds and attenuations for the larger grain sizes (phi < -1) need to be updated, and the latter that the sediment densities for the middle range of grain sizes (1 < phi < 5) are underestimated. On the last point, the authors of the original model were aware of the problem, and for practical reasons decided to accept the understatement in the interests of achieving the correct reflection loss. The discrepancies may be alleviated by adopting a poro-elastic model with multiple scattering corrections. For practical applications, an efficient parameterization of the poro-elastic model allows the number of adjustable parameters to be reduced to a level comparable with that of simpler fluid and elastic models, while retaining all its physical advantages. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]



**2:00**

**4pUW3. Measurements of compressional wave dispersion and gradients in muddy sediments.** Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Jan Dettmer, Stan Dosso, and Gavin Steininger (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Cohesive or muddy sediments have received relatively sparse attention in the ocean acoustics community—this despite the fact that they form a non-negligible fraction of the sediments found in shallow water (roughly 25%) and by far the major sediment type in the deep ocean. Seabed reflection measurements have provided some understanding about the frequency dependence of the sound speed and attenuation in muddy sediments. The evidence is for weak dispersion from 300–200,000 Hz and an approximately linear dependence of

attenuation on frequency from 300–3000 Hz. In addition, the measurements have yielded information on gradients. Surprisingly large near-surface density gradients exist that vary across the shelf. Given the large density gradients, the gradients in sound speed are curiously small, suggesting that the bulk modulus is nearly proportional to the density, at least in depth. Dispersion and gradient results are discussed for muddy sediments in various mid to outer shelf regions.

## *Contributed Papers*

**2:20**

**4pUW4. Issues in reverberation modeling.** Dajun Tang (Appl. Phys. Lab., Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

Reverberation usually consists of two-way propagation, or forward scatter, and a single backscatter. The scattering cross section is often employed to couple the two-way propagation to obtain approximate reverberation strength. Because this approach is inherently incoherent and heuristic in nature, certain limitations to its applicability need to be elucidated. In particular, unlike backscatter problems in half-space, reverberation in shallow water involves coherent incident fields at different wavenumbers. Starting with the fundamental definition of scattering T-matrix and through examples, this paper intends to address the following issues: (1) how to incorporate coherent component of reverberation into simulations, (2) how to rigorously relate time to range for given bandwidth, and (3) how to increase computation speed through proper smoothing. [Work supported by ONR.]

**2:35**

**4pUW5. Seismic sources in seismo-acoustic propagation models.** Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, 1500 Illinois St., Golden, CO 80401, jcollis@mines.edu), Scott D. Frank (Mathematics, Marist College, Poughkeepsie, NY), Adam M. Metzler (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Robert I. Odom (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

An important generating mechanism for received underwater acoustic and seismic signals are buried or earth-bound sources. Most underwater acoustic studies involve purely compressional sources in the water column. The more complicated case of a coupled shear and compressional seismic source in the sediment has recently been implemented in an elastic parabolic equation solution [Frank *et al.*, J. Acoust. Soc. Am. **133**]. In this talk, generic seismic sources including those giving shear field contributions, are contrasted in normal mode and parabolic equation solutions. Scenarios considered are for an elastic-bottom Pekeris waveguide and a canonical Arctic propagation scenario with an elastic ice cover over the ocean and an elastic basement. For the Arctic case, the source is allowed in either the ice cover or in the elastic bottom. Solutions are benchmarked for purely compressional and shear seismic sources, and their relation to the seismic moment tensor is discussed. The ultimate goal of these solutions is to allow for seismic sources capable of representing generic geophysical events.

**2:50**

**4pUW6. Nonlinear acoustic pulse propagation in dispersive sediments using fractional loss operators.** Joseph T. Maestas (Appl. Mathematics and Statistics, Colorado School of Mines, 1500 Illinois St., Golden, CO 80401, jmaestas@mines.edu)

The nonlinear progressive wave equation (NPE) is a time-domain formulation of Euler's fluid equations designed to model low-angle wave propagation using a wave-following computational domain [ McDonald *et al.*, J. Acoust. Soc. Am. **81**]. The wave-following frame of reference permits the simulation of long-range propagation that is useful in modeling the effects of blast waves in the ocean waveguide. However, the current model does not take into account sediment attenuation, a feature necessary for accurately describing sound propagation into and out of the ocean sediment. These attenuating, dispersive sediments are naturally captured with linear, frequency-domain solutions through use of complex wavespeeds, but a comparable treatment is nontrivial in the time-domain. Recent developments in fractional loss operator methods allow for frequency-dependent loss mechanisms to be applied in the time-domain providing physically realistic results [Prieur *et al.*, J. Acoust. Soc. Am. **130**]. Using these approaches, the governing equations used to describe the NPE are modified to use fractional derivatives in order to develop a fractional NPE. The updated model is then benchmarked against a Fourier-transformed parabolic equation solution for the linear case using various sediment attenuation factors.

**3:05**

**4pUW7. A scaled mapping approach for range-dependent seismo-acoustic propagation using the parabolic approximation.** Adam M. Metzler (Appl. Res. Labs., The Univ. of Texas at Austin- ARL-Environ. Sci. Group, PO Box 8029, Austin, TX 78713, ametzler@arlut.utexas.edu), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Parabolic equation solutions are used to accurately and efficiently model range-dependent propagation effects in ocean environments. There has been much recent interest in improving accuracy, particularly for sloping interfaces between fluid and underlying sediment layers. A translational mapping approach [Collins *et al.*, J. Acoust. Soc. Am. **107** (2000)] applies a coordinate transformation in which a sloping bottom interface becomes horizontal and range dependence is mapped to the upper free surface. While accurate for small slopes, this approach introduces errors for variably sloping bathymetries since the range dependence is transformed to the surface. In this work, a scaled mapping is constructed that both transforms the sloping bottom interface to horizontal and also preserves the range-independent form of the free surface by distorting the waveguide in depth. The parabolic approximation is applied in the fully range-independent transformed domain, and the result is inverse transformed to obtain the solution in the initial range-dependent environment. Applications of this approach are given and benchmarked for seismo-acoustic propagation scenarios. [Work supported by ARL:IR&D.]

**3:20**

**4pUW8. Volume scattering and reverberation in shallow water: A simplified modeling approach.** Anatoliy Ivakin (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

A simplified physics-based approach is described that allows significantly faster yet reasonably accurate estimations of volume reverberation in complex shallow water environments. An integral expression is presented for scattering intensity with a factorized integrand comprised of two kernels, the double propagator and local volume scattering coefficient. The propagator describes the local intensity and can be calculated using available models, such as PE, normal modes, or ray approximations. The scattering kernel can be specified using available volume scattering models for continuous or discrete heterogeneity of sea-water column and seabed caused by spatial fluctuations of compressibility and density, or randomly distributed discrete targets, such as bubbles, fish, shells, and others. The approach is more general than and can be used for verification of existing reverberation models. For instance, calculation of bottom reverberation is not based on using the equivalent surface scattering strength (although considers it as a particular case). Numerical examples for shallow water reverberation time series, based on a PE propagation model, are presented to estimate potential contributions of different mechanisms of scattering. The estimations provide a comparison of relative contributions of scatterers with the same scattering strengths but located at different depths in water column or in the sediment. [Work supported by the US Office of Naval Research.]

**3:35**

**4pUW9. Computation of the field of coupled modes using split-step algorithm.** Nikolai Maltsev (R&D, Frontier Semiconductor, 2127 Ringwood Ave., San Jose, CA 95131, admin@asymptotus.com)

Euler equations in the form $\partial\mathbf{F}/\partial x = \mathbf{A}(x,z)\mathbf{F}$ where $2\times 2$ matrix $\mathbf{A}$ has elements $A_{11}=0$, $A_{22}=0$, $A_{12}=i\omega\rho$, $A_{21}=1/(i\omega\rho)(-\triangle_{yz}+(\nabla_{yz}\ \ln\rho,\nabla_{yz})-(\omega/c)^2)$ where $\mathbf{F}=(P(\mathbf{r}),\ u(\mathbf{r}))^T$ are sound pressure and horizontal velocity, $c(\mathbf{r})$, $\rho(\mathbf{r})$ -sound speed and density, $\omega = 2\ \pi f$—angular frequency and $\triangle_{yz},\nabla_{yz}$ are Laplace operator and gradient in the plane (y,z), has first order with respect to x and can be integrated by split step algorithm $\mathbf{F}(x+d)=\exp(0.5\mathbf{A}(x+d)d)\exp(0.5\mathbf{A}(x)d)\mathbf{F}(x)+O(d^3)$ using local modes for computation of exponential operators. Integration is performed in one direction but, due to the structure of normal modes of operator $\mathbf{A}$, allows estimate energy reflected back on every integration step. Different examples, including irregular waveguides with ideal boundaries and Pekeris style guide with variable depth are presented.

THURSDAY EVENING, 5 DECEMBER 2012 — 8:00 P.M. TO 10:00 P.M.

## OPEN MEETINGS OF TECHNICAL COMMITTEES

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

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

Committees meeting on Thursday are as follows:

| | |
|---|---|
| Musical Acoustics | Continental 1 |
| Speech Communication | Plaza B |
| Underwater Acoustics | Golden Gate 2/3 |

## Session 5aAB

# Animal Bioacoustics: Animal Hearing and Vocalization

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

### *Contributed Papers*

**9:00**

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

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

**9:15**

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

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

**9:30**

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

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

**9:45**

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

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

10:00

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

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

10:15

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

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

10:30

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

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

## Session 5aBAa

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

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

Gail ter Haar, Cochair
*Phys., Inst. of Cancer Res., Royal Marsden Hospital, Downs Rd., Sutton SM25PT, United Kingdom*

**Chair's Introduction—7:55**

### *Invited Papers*

**8:00**

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

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

**8:20**

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

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

**8:40**

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

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

**9:00**

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

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

## *Contributed Papers*

**9:20**

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

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

**9:35**

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

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

**9:50**

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

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

**10:05**

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

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

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

**10:20–10:30 Break**

**10:30**

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

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

**10:45**

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

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

**11:00**

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

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

**11:15**

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

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

**11:30**

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

B-scan imaging relies on geometric focusing of ultrasound echoes. In contrast, the method of Shear Wave Induced Phase Encoding imaging achieves lateral resolution of an elastic target by using traveling shear waves to encode the position of scatters in the phase of the received echo. A Fourier series description of the phase modulated echo signal is developed, demonstrating that echo harmonics at multiples of the shear wave frequency

reveal target k-space data at identical multiples of the shear wavenumber. Modulation transfer functions (MTFs) are calculated for maximum shear wave acceleration and maximum shear constraints, and compared with a conventionally focused aperture. The relative SNR of this method versus a conventionally focused aperture is determined from these MTFs. Reconstructions of wire targets in a gelatin phantom using a cylindrical shear wave source are presented, including Images generated from the fundamental and second harmonic of the shear wave modulation frequency. Comparison of images generated at 1 and 3.5 MHz demonstrate the weak dependence of lateral resolution on ultrasound frequency with this method.

**11:45**

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

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

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

# Session 5aBAb

## Biomedical Acoustics: High Frequency Ultrasound

Jeffrey A. Ketterling, Chair
*Riverside Res., 156 William St., New York, NY 10038*

### *Contributed Papers*

**8:00**

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

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

**8:15**

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

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

8:30

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

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

8:45

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

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

9:00

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

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

9:15

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

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

9:30

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

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

9:45

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

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

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

## Session 5aEA

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

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

### *Contributed Papers*

8:30

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

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

8:45

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

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

9:00

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

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

9:15

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

The acoustic performance of exhaust systems is affected by the flow speed and temperature of the exhaust gases. The flow speed affects the convective wave speed and changes the acoustic properties of some acoustic elements such as perforates. The temperature of the exhaust gas affects its density and the speed of sound. It is important to model the flow and temperature distribution within the exhaust system in order to perform an accurate acoustic simulation. The acoustic propagation along an exhaust system can be modeled by dividing the exhaust system into a number of two-port

elements. It has been previously shown that flow distribution and pressure drop calculation can be done using the same two-port elements but using a different set of transfer matrices. A similar approach was previously proposed to calculate the temperature drop using the same two-port elements but using a new set of transfer matrices. This technique has the advantage to solve all physics by defining only one network. An improved version of this formulation is presented in this paper. Several cases were tested and the thermal and acoustic results were compared to finite element simulations and measurements. The two port results matched well with the other techniques.

**9:30**

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

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

**9:45**

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

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

**10:00–10:15 Break**

**10:15**

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

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

**10:30**

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

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

**10:45**

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

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

## Session 5aMU

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

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

### *Invited Papers*

**10:00**

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

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

**10:20**

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

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

**10:40**

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

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

**11:00**

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

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

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

**11:20**

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

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

**11:40**

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

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

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

# Session 5aNS

## Noise: Perception and Measurement of Sound and Sound and Equipment

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

### *Contributed Papers*

**8:00**

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

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

**8:15**

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

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

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

**8:30**

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

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

**8:45**

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

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

**9:00**

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

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

**9:15**

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

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

**9:30**

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

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

**9:45**

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

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

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

**10:00–10:15 Break**

**10:15**

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

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

**10:30**

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

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

**10:45**

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

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

**11:00**

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

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

## Session 5aPAa

## Physical Acoustics: Advances in Infrasound Research III

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

### *Invited Papers*

**8:00**

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

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

**8:20**

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

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

**8:40**

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

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

**9:00**

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

The performance of an infrasound array is typically described qualitatively. In this project, we begin determining quantitative metrics that define the performance of an infrasound array based upon changes in array parameter estimation, particularly direction-of-arrival and trace velocity. First, we determine the magnitude and frequency response of an International Monitoring System (IMS) infrasound array using the *in-situ* calibration technique. Phase differences in the frequency response between the reference and actual array elements are then represented as a shift in time. These times shifts are used to calculate time differences between non-redundant sensor pairs of the array and are then compared to theoretical time differences for incident plane waves. This allows us to estimate the change in direction-of-arrival and trace velocity due to the unique frequency response of an array using a least-squares technique. We present detailed results for IMS arrays IS26, IS53, IS56, and IS57, which provides a good sampling of different sensors and wind-noise reduction systems in the IMS. For IS53 (Fairbanks, AK), the deviation in the azimuth estimation is bounded below 2 degrees at all frequencies of interest (>0.02 Hz). Preliminary results suggest array parameter estimation at other arrays can be significantly affected by changes in frequency response.

**9:20**

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

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

**9:40**

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

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

**10:00–10:15 Break**

## *Contributed Papers*

**10:15**

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

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

**10:30**

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

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

**10:45**

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

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

**11:00**

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

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

**11:15**

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

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

5a FRI. AM

## Session 5aPAb

# Physical Acoustics: Topics in Physical Acoustics

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

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

### *Contributed Papers*

**8:00**

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

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

**8:15**

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

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

**8:30**

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

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

**8:45**

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

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

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

**9:00**

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

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

**9:15**

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

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

**9:30**

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

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

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

## Session 5aPP

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

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

### *Contributed Papers*

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

# Session 5aSCa

## Speech Communication: Predictive Processing

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

### *Contributed Papers*

**8:30**

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

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

**8:45**

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

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

**9:00**

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

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

**9:15**

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

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

9:30

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

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

**9:45–10:00 General Discussion**

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

# Session 5aSCb

## Speech Communication: Neurophonetics

Marc Ettlinger, Chair

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

### *Contributed Papers*

10:15

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

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

10:30

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

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

### 10:45

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

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

### 11:00

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

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

### 11:15

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

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

**11:30–11:45 General Discussion**

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

## Session 5aSCc

## Speech Communication: Speech Analysis (Poster Session)

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

### *Contributed Papers*

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

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

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

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

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

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

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

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

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

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

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

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

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



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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Voice activity sensors commonly measure voiced-speech-induced skin vibrations using contact microphones or related techniques. We show that micro-power EM wave sensors have advantages over acoustic techniques by directly measuring vocal-fold motions, especially during closure. This provide 0.1 ms timing accuracy (i.e., ~10 kHz bandwidth) relative to the corresponding acoustic signal, with data arriving ~0.5 ms in advanced of the acoustic speech leaving the speaker's mouth. Preceding or following unvoiced and silent speech segments can then be well defined. These characteristics enable anti-speech waves to be generated or prior recorded waves recalled, synchronized, and broadcast with high accuracy to mask the user's real-time speech signal. A particularly useful masking process uses an acoustic voiced signal from the prior voiced speech period which is inverted, carefully timed, and rebroadcast in phase with the presently being spoken acoustic signal. This leads to real-time cancellation of a substantial fraction of the voiced acoustic energy, as well as providing timing to mask the remaining un-canceled voiced speech energy, and unvoiced speech and silence periods.

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

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

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

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

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

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

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

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

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

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

## Session 5aUW

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

Nicholas P. Chotiros, Cochair
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**Chair's Introduction—7:55**

### *Invited Papers*

**8:00**

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

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

**8:20**

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

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

**8:40**

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

During the Target and Reverberation Experiment 2013 (TREX13), high-frequency measurements of sediment sound speed and attenuation were collected throughout the experiment site. These measurements were performed using the INSEA, a diver-portable array of sources and receivers conceived and developed by French companies in collaboration with research institutions. During each deployment of the instrument, the INSEA was inserted 10–15 cm into the sediment and narrow-band pulses covering the 70 to 350 kHz range

were transmitted through the sediment. The sound speed is determined from the time-of-flight and attenuation is determined from the amplitude ratio of the transmissions through the sediment and through the water. The variability of the TREX13 site made it possible to collect data in several different sediment types including mud, silty-sand, and sand sediments each with low to high concentrations of shells. In addition to the acoustic measurements, diver cores and shell samples were also collected. The sound speed and attenuation measured in these sediments are presented and discussed. [Work supported by DGA, ONRG, and SERDP.]

## *Contributed Papers*

**9:00**

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

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

**9:15**

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

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

**9:30**

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

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

**9:45**

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

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

**10:00–10:15 Break**

**10:15**

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

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

**10:30**

**5aUW9. Reflection-coefficient inversion for compressional- and shear-wave dispersion in porous and elastic layered media.** Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8W 3P6, Canada, jand@uvic.ca), Charles W. Holland (Appl. Res. Lab., The Penn State Univ., State College, PA), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

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

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

**10:45**

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

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

**11:00**

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

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

**11:15**

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

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

## Session 5pMU

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

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

### *Invited Papers*

**1:00**

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

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

**1:20**

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

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

**1:40**

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

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

### *Contributed Papers*

**2:00**

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

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

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

**2:15**

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

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

**2:30**

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

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

**2:45**

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

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

**3:00**

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

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

**3:15**

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

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

**3:30**

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

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

## Session 5pSC

## Speech Communication: Crosslinguistic Analysis (Poster Session)

Molly E. Babel, Chair

*Linguist., Univ. of British Columbia, 2613 West Mall, Totem Field Studios, Vancouver, BC V6T 1Z4, Canada*

### *Contributed Papers*

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

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

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

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

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

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

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

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

T1 was lower than other contrasts with the exception of T2-T3. Both Japanese and Thai listeners had greatest difficulty with T2-T3. Since previous knowledge of L1 tones may interfere with the perception of non-native tones, these results will be discussed with reference to a Thai tonal system.

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

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

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

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

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

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

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

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

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

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

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

The purpose of the present study is to examine word-initial stop consonants of Canadian English (CE) and Canadian French (CF) interlingual homophones in order to describe how they differ in their acoustic properties. Interlingual homophones (IH) are words across languages that are phonemically identical but phonetically and semantically different, for example, English two /tu/ and French tout <all> /tu/. Even though they are deemed phonemically identical, at the acoustical level they may be quite different. In the current study, Canadian bilingual English and French speakers were asked to produce interlingual homophones embedded in carrier phrases and in isolation. Voice onset time, relative burst intensity, and burst spectral

properties of the IH words were measured and compared within and across languages. The acoustic measurements obtained will be used (1) to make predictions about which acoustic features may provide cues to language identity, and (2) to create stop tokens for a Goodness Rating study. Results from this study will provide insight on the acoustic-phonetic representation of stop consonants in Canadian bilingual English and French speakers.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

brain structure between monolinguals and bilinguals as consistent with the idea that bilingual language experience leads to a pattern of more diffuse connectivity in the brain, which may be related to increased cognitive flexibility skills.

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

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

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

## Session 5pUW

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

Nicholas P. Chotiros, Chair
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**Chair's Introduction—1:00**

### *Invited Papers*

**1:05**

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

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

**1:25**

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

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

1:45

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

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

## *Contributed Papers*

2:05

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

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

2:20

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

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

2:35

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

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

2:50

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

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

3:05

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

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