Session 4aAA

Architectural Acoustics and ASA Committee on Standards: Acoustics of Mixed Use Buildings

Steven D. Pettyjohn, Chair
The Acoustics & Vibration Group, 5700 Broadway, Sacramento, CA 95820-1852

Chair's Introduction—8:00

Invited Papers

8:05

4aAA1. Acoustical design of mixed-use buildings: The state of the art (and science). Robert P. Alvarado and Ethan C. Salter (Charles M. Salter Assoc., Inc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, robert.alvarado@cmsalter.com)

Mixed-use developments integrate a variety of uses including residences, offices, shops, restaurants, and theaters into functional, living, and working communities. They are often located near rail, major roadways, and airports, which can generate significant levels of environmental noise and vibration that will need to be mitigated to meet Building Codes and project goals. This paper will detail project-specific issues, solutions, and experiences by Charles M. Salter Associates. Our experience has given us the opportunity to collaborate with various project stakeholders to achieve project goals within the constraints of aesthetics, budget, and space. Authors will present case studies which include acoustical design issues that have been incorporated during the design phase, coordinated during construction, and conveyed to the community that the project serves. Acoustical needs of mixed-use developments can vary based on applicable Codes, as well as project marketing goals and expectations of end users. Educating future owners and tenants about acoustics is important to reducing the design team and the developers' exposure to possible litigation, and also helps to maintain resident health, comfort, and property values. Robert Alvarado has managed hundreds of projects over the past 12 years. His mixed-use project portfolio includes international and domestic projects. Ethan Salter has consulted on dozens of projects throughout the country, including several incorporating sustainable design.

8:25

4aAA2. From an architect's perspective: Acoustical challenges in mixed-use buildings. Dale Farr and Karyn Goodfriend (Fletcher Farr Ayotte, Inc., 520 SW Yamhill, Ste. 900, Portland, OR 97204, kgoodfriend@ffadesign.com)

Incorporating acoustical design in mixed-use buildings can be architecturally challenging. Key issues impacting acoustical design decisions are: (1) the unknown—due to undetermined uses of commercial lease space; (2) the cost—clients do not want to pay for ideal acoustical design because they perceive it to be cost prohibitive. Often clients do not understand the importance of considering acoustic elements during the initial design process until complaints occur much later, postconstruction. Other elements impacting acoustical design are: apartments versus condominiums, existing buildings versus new structures; changes in lease space use; and the possible conflict of the acoustical design not meeting life/safety code. On each architectural project it is ultimately the goal to design to client criteria with a mission to deliver solutions that meet the most needs based on user programs. Often concessions must be made to find the best overall solution for the building and budget. This impacts all of the design consultants, which means designs need to be flexible and are sometimes less than ideal in segregation. It is worth discussing how architects successfully design for the unknown and justify inclusion of acoustical designers early on, while staying within budget.

8:45

4aAA3. Sound and vibration issues encountered in mixed use construction involving residential, office, and retail uses created from a combination of renovation and expansion of an existing space. Steven D. Pettyjohn (The Acoust. & Vib. Group, Inc., 5700 Broadway, Sacramento, CA 95820-1852, spettyjohn@acousticsandvibration.com)

Buildings housing mixed uses often encounter acoustic and vibration problems because expectations differ for the users and patrons. Mixed uses could include retail and office spaces; offices and residential; retail and residential; or, as in the case to be discussed, retail (restaurants) on the first floor, offices on the second floor, and residences on the upper two floors. An existing historic three-story concrete structure originally housing a car dealership in an urban area was renovated to provide all three uses. A fourth floor was added to provide a second floor of residential use. The offices and residential areas were designed to meet interior sound level standards from sources within and exterior to the building. The two restaurants were designed individually by their in house teams with only vibration standards provided by the building design group. The lack of coordination between restaurant and building design teams let to both sound and vibration problems that had to be corrected after construction was mostly complete. Construction teams for the restaurant groups were not familiar with such issues based on their previous installations. This paper discusses the myriad of problems and remedies and provides an outline of the issues that need addressing.

9:05

4aAA4. Low frequency sound problems found in mixed use buildings that house entertainment venues and residential developments and containment options. Scott W. Smith (Ballentine Walker Smith, Inc., Kennesaw, GA 30144, bwsacoustics@bellsouth.net) and Steven D. Pettyjohn (The Acoust. & Vib. Group, Inc., Sacramento, CA 95820-1852)

Many problems are to be expected when mixed use buildings include restaurants and residential spaces. When the residential spaces are condominiums and the restaurant becomes a nightclub, the sound problems multiply quickly. The low frequency sound produced in a nightclub featuring music catering to a young crowd is of particular concern. This is partially because of the difficulty of finding remedies once the building construction is complete. This is the situation that arose in a facility recently completed. The nightclub wanted to continue its operation while the condominium owners wanted a resolution of the problem. Sound tests were completed in the residential spaces during operation of the nightclub, but the low frequency content was not always the same, requiring multiple attempts to measure in the source and receiving spaces. Results of these measurements and the recommendations for correcting the problem are presented in this paper. The goal is to provide results of sound measurements made after the recommendations are implemented. Again, sound will be measured in the source and receiver spaces to understand how the noise reduction changed and compared with the predicted sound reduction.

Contributed Paper

9:25

4aAA5. An historic conversion: From a bank to a restaurant and residences. Ioana Pieleanu, Jeffrey Fullerton, and Benjamin Markham (Acentech Inc., 33 Moulton St., Cambridge, MA 02135)

A conversion of an old bank building in Boston's tony South End to a mixed-use building featuring retail on the ground floor and luxury condominiums above was completed in 2007. More recently, a new restaurant (garnering awards for its interior design and rave reviews for its food) has opened in one of the ground floor retail spaces directly below a particularly noise-sensitive resident. Consultants at Acentech worked on two aspects of the project: first, on the base building as consultants to the architect, and second, on the isolation between the restaurant and the second floor residences as consultants to the restaurant. Using this case study and extensive data measured on site, the authors will discuss best practices to achieve good sound isolation in mixed-use buildings, common pitfalls that result from working with existing historic structures, and some difficulties in achieving the high degree of sound isolation that some luxury condominium owners expect.

THURSDAY MORNING, 21 MAY 2009 GALLERIA NORTH, 8:25 TO 11:50 A.M.

Session 4aAB

Animal Bioacoustics: General Topics in Animal Bioacoustics I

Holger Klinck, Chair
CIMRS, Oregon State Univ., Newport, OR 97365

Chair's Introduction—8:25

Contributed Papers

8:30

4aAB1. Auditory temporal summation in pinnipeds. Asila Ghoul (Univ. of California Santa Cruz Long Marine Lab., 100 Shaffer Rd., Santa Cruz, CA 95060), Marla M. Holt (Natl. Marine Fisheries Service, Seattle, WA 98112), Colleen Reichmuth, and David Kastak (Univ. of California Santa Cruz Long Marine Lab., Santa Cruz, CA 95060)

In addition to improving the understanding of auditory processing in pinnipeds, direct measures of temporal summation are relevant to the selection of signal parameters when conducting audiometric research, assessing the effects of signal duration on communication ranges, and evaluating the potential auditory impacts of anthropogenic signals. In the present study, individuals from three pinniped species were tested to determine how signal duration influenced pure-tone hearing thresholds. The psychophysical method of constant stimuli was used to obtain aerial thresholds for each subject at nine different signal durations ranging from 25 to 500 ms. Parameter estimates derived for a California sea lion (*Zalophus californianus*) from an exponential model of temporal summation yielded time constants (τ) of 176, 98, and 141 ms at frequencies of 2.5, 5, and 10 kHz, respectively. Preliminary results with a northern elephant seal (*Mirounga angustirostris*) at 5 kHz (this study), and a harbor seal (*Phoca vitulina*) at 2.5 kHz [M. M. Holt et al., J. Soc. Am. **116**, 2531 (2004)] show similar values for (τ), 134 and 144 ms, respectively. These time constants are similar to those of other mammals tested and do not appear to vary with respect to frequency.

8:45

4aAB2. Annual temporal patterning in the vocalizations of captive seals: Two long-term case studies. Colleen Reichmuth and Ronald J. Schusterman (Inst. of Marine Sci., Univ. of California Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060)

Seasonal changes in vocalizations occur in a variety of species. Factors such as the condition of conspecifics, physiological states that in turn may be related to environmental cues, and developmental and individual differences all potentially influence temporal changes in sound production. In the present study, the vocal behavior of two captive seals was monitored daily for over 10 yrs. Both seals were housed in the absence of conspecifics from the age of 1 yr extending past sexual maturity. The male harbor seal (*Phoca vitulina*) began characteristic underwater vocal displays at the age of 6. Intense periods of acoustic activity lasted weeks to months, overlapped with the breeding activity of local harbor seals, and comprised stereotypic sound emissions that were structurally similar to those reported for wild seals. The female northern elephant seal (*Mirounga angustirostris*) produced aberrant intense airborne vocalizations from the age of 4 that were annually synchronized to a period of approximately 5 weeks coinciding with estrous. Endogenous changes appear to trigger these behavioral cycles, presumably as a

result of hormonal changes associated with photoperiod. Vocalizations may be a noninvasive indicator of reproductive state and therefore may provide a useful management and conservation tool in captive settings.

9:00

4aAB3. A comparison of behavioral and electrophysiological measures of aerial hearing sensitivity in a Steller sea lion (*Eumetopias jubatus*). Jason Mulsow (Dept. of Ocean Sci., Univ. of California Santa Cruz, Earth and Marine Sci. Bldg., Santa Cruz, CA 95064) and Colleen Reichmuth (Univ. of California Santa Cruz, Santa Cruz, CA 95060)

A number of studies with odontocete cetaceans have demonstrated that hearing sensitivity measurements using electrophysiological auditory steady-state responses (ASSRs) can provide an efficient means of estimating a subject's behavioral audiogram. Expansion of ASSR methods to another marine mammal group, the otariid pinnipeds (sea lions and fur seals), holds the potential to increase the number of otariid individuals and species for which hearing sensitivity data are available. A within-subject comparison of ASSR and behavioral measures of aerial hearing sensitivity was conducted with an individual of the largest otariid species, the Steller sea lion. Psychophysical methods were used to obtain an unmasked aerial audiogram at 13 frequencies spanning a range of 0.125 to 34 kHz. Corresponding ASSR thresholds measured at frequencies of 1, 2, 5, 10, 20, and 32 kHz had differences (relative to behavioral thresholds) ranging from 1 dB at 20 kHz to 30 dB at 1 kHz. Overall, the ASSR audiogram was a fairly accurate predictor of the behavioral audiogram at frequencies of 2 kHz and above. Our results suggest that ASSR methods can be appropriately applied to otariid pinnipeds in estimating aerial sensitivity at frequencies of approximately 2 kHz and above.

9:15

4aAB4. Vibration characteristics of the tympanoperiotic complex in the bottlenose dolphin, *Tursiops truncatus*. Petr Krysl (Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0085), Ted W. Cranford (San Diego State Univ., San Diego, CA 92182), and John A. Hildebrand (Univ. of California, San Diego, La Jolla, CA 92093-0205)

Modal finite + boundary element analysis of a bottlenose dolphin's bony tympanoperiotic complex, including the ossicles, was performed to determine the mode shapes and natural frequencies. The goal was to gain insight into the transmission of sound pressure waves arriving through the soft tissues and transmitted across the bony components into the oval window of the inner ear. The finite element model of the bones was derived from CT scans with a 360 μm voxel resolution. In the first approximation the soft tissue was considered to be acoustically equivalent to an incompressible inviscid liquid, taken as infinite in extent. The added mass terms were computed with a boundary element model. The computed frequencies cover the range up to 160 kHz. The capacity of the natural vibration modes to excite motion of the stapes footplate was assessed by measuring the relative motion of the incudostapedial joint normalized by the normal displacement of the wet-surface of the ear bones. In addition to a quantitative assessment a number of qualitative observations may be made that could explain the function of the dolphin's ear complex. For example, the vibrational patterns are nontrivial and frequency dependent. [Work supported by the U.S. Navy CNO45.]

9:30

4aAB5. "Rivers" of sound in Cuvier's beaked whale (*Ziphius cavirostris*): Implications for the evolution of sound reception in odontocetes. Ted W. Cranford (Biology Dept., San Diego State Univ., 2674 Russmar, San Diego, CA 92182), Petr Krysl, and John A. Hildebrand (Univ. of California at San Diego, La Jolla, CA 92093)

Industrial CT scanning technology was used to collect the first x-ray tomograms from the head of an adult male Cuvier's beaked whale. These scans and tissue property measurements were used to construct a finite element model. Simulations revealed pathways for sound propagation into and out of the head. One intriguing result concerns a newly described gular pathway by which sound reaches the hearing apparatus. Propagated sound waves enter the ventral aspect of the head and form an acoustic "river" that flows toward the bony ear complexes through the internal mandibular fat bodies. The precise pathway and dimensions of the sound river vary with frequency, but it converges on the bony tympanoperiotic complex. A combination of tissue structures and air spaces act like an internal acoustic pinna that filters and concentrates the incoming sound. The river of sound apparently functions in concert with the absence of the medial bony lamina of the posterior portion of the mandible, a condition that exists in all toothed whales and their ancestral archaeocetes. The gular pathway and river of sound suggests that this is the primordial pathway for underwater hearing in whales and that Norris' jaw hearing mechanism was a more recent development.

9:45

4aAB6. Dall's porpoise (*Phocoenoides dalli*) echolocation click spectral structure. Hannah R. Bassett, Simone Baumann, Gregory S. Campbell, Sean M. Wiggins, and John A. Hildebrand (Marine Physical Lab, Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, hbassett@ucsd.edu)

Dall's porpoise (*Phocoenoides dalli*) echolocation clicks have not been widely recorded. Concurrent with visual observations, acoustic recordings of free-ranging Dall's porpoise were made offshore of southern California using a towed hydrophone array with two elements of 250 kHz bandwidth. We examined 6035 clicks from 12 sessions totaling more than two hours over the course of seven days. The Dall's porpoise echolocations recorded were short (48–804 μs), narrow band (2–10 kHz [−3dB]) clicks with most peak frequencies between 117 and 141 kHz, but some as high as 198 kHz. Many clicks contained a multipulse temporal structure, resulting in stereotyped spectral peaks and notches. Two distinctive click types with different spectral banding patterns and peak frequencies (122.8 and 135.8 kHz) were observed. Spectral banding patterns have been used as a species identifier for Risso's dolphins and Pacific white-sided dolphins. These two dolphins and Dall's porpoise have similar head morphologies, which may play a role in producing clicks with spectral peaks and notches. This study shows that Dall's porpoise produce multiple click types, which may provide a tool for population classification, and that their clicks contain spectral banding patterns, which may provide insight into the mechanism by which such clicks are produced.

10:00—10:20 Break

10:20

4aAB7. Analysis of most prominent signal features of humpback whale (*Megaptera Novaeangliae*) vocalizations towards the goal of autonomous acoustic classification. Ted Abbot, Owen Mayer, Vince Premus, Philip Abbot, and Ira Dyer (OASIS, Inc., 5 Militia Dr. Lexington, MA 02421)

Humpback whale vocalizations were recorded using hydrophones on glider systems off Alaska in January 2000, in Hawaii in February 2008, and in the Stellwagen Bank National Marine Sanctuary in October 2007 and July 2008. The vocalizations have been grouped into five call types based on the most prominent signal features. Only five call types are used because autonomous species classification relies on the most consistent and repeatable signal features rather than the full diverse range of humpback vocalizations. The five call types are upsweep (increasing frequency over time), downsweep (decreasing frequency over time), flute (increasing and decreasing frequency over time), tone (little or no change in frequency over time), and groan (commonly a social or feeding-related vocalization, frequently characterized by unstructured broadband sound). We present detailed statistical analyses of these call types including bandwidth, minimum and maximum frequency, duration, and slope. A comparative analysis across data sets shows the relative frequency of occurrence of each vocalization type and indicates the degree of temporal and geographic variation of Humpback vocalizations.

10:35

4aAB8. Is rejection of clutter achieved by disrupting perception of delay in bat sonar? Mary E. Bates (Dept. of Psych., Brown Univ., 89 Waterman St., Providence, RI 02912) and James A. Simmons (Brown Univ., Providence, RI 02912)

Big brown bats (*Eptesicus fuscus*) emit frequency-modulated (FM) biosonar sounds containing two prominent harmonics, FM1 (55–22 kHz) and FM2 (105–145 kHz), and perceive target distance from echo delay. Ordinarily, echoes from objects arrive with both harmonics at the same delay, although FM2 becomes progressively more attenuated than FM1 with increasing target distance and off-axis direction. Off-axis or long-range echoes naturally undergo lowpass filtering, which causes significant loss of acuity. Misalignment of FM2 with respect to FM1 or selective removal of low-end frequencies disproportionately affects the accuracy of delay perception. Delay acuity is sharp for echoes containing both harmonics, less sharp for echoes containing only FM1, and very poor for echoes containing only FM2. Acoustically unnatural highpass filtering to remove low-end frequencies from FM1 causes acuity to collapse. Attenuation of FM2 relative to FM1 by 3 dB decreases delay acuity, but shortening of the delay for FM2 by 48 μs counteracts amplitude-latency trading and restores delay acuity. Bats may have a spatial perceptual "fovea" covering a narrow zone in front of them for high-acuity perception surrounded by a zone of much lower acuity that suppresses the perceptual salience of background clutter. [Work supported by NASA RI Space Grant and ONR.]

10:50

4aAB9. Range discrimination of multiple objects from the echo spectrogram measured by using the frequency modulation sound. Ikuo Matsuo (Dept. of Information Sci. Tohoku Gakuin Univ., Sendai, 981-3193, Japan, matsuo@cs.tohoku-gakuin.ac.jp)

Using the echolocation, bats can capture moving objects in real 3D space. Bats emit the frequency modulation sound and can identify objects with an accuracy of less than a millimeter. To determine delay times of multiple objects requires estimating the sequence of delay separations by extracting temporal changes in the interference pattern of the echoes. The models to determine delay times of multiple objects from the simulated echoes by using the frequency modulation sound have been previously proposed. In order to extract the temporal changes, Gaussian chirplets with a carrier frequency compatible with emission sweep rates were used. The delay time for first object can be estimated from the echo spectrum around the onset time. The delay time for second object is obtained by adding the delay time for the first object to the delay separation between first and second objects (extracted from the first appearance of interference effects). Further objects can be located in sequence by this same procedure. In this paper, these echoes were measured from multiple closely spaced objects, and it was examined that this model could estimate each delay times of these objects.

11:05

4aAB10. The acoustic implications of sella length in horseshoe bats. Rolf Mülller (Dept. of Mech. Eng., Virginia Tech. & Inst. for Adv. Learning and Res., 150 Slayton Ave., Danville, VA 24540, rolf.mueller@vt.edu), Zhiwei Zhang (Shandong Univ., Hongjia Lou 5, 250100 Jinan, China), and Son Nguyen Truong (Vietnamese Acad. of Sci. and Technol., Hanoi, Vietnam)

Horseshoe bats emit their biosonar pulses through nostrils that are surrounded by elaborate baffle shapes (noseleaves). The shape and size of the different noseleaf shape elements can vary considerably between species. Here, the acoustic effects of the length of the sella have been investigated based on the noseleaf of Bourret's Horseshoe Bat (*Rhinolophus paradoxolophus*) which features an exceptionally long sella: High-resolution digital representations of the noseleaves of three specimens were used to vary the sella length below and above its natural value. A principal axis through the sella was used to define the direction of this scaling. The acoustic beamforming properties of the noseleaf were then assessed as a function of sella scale using numerical methods. It was found that elongating the sella narrows the biosonar beam of the second harmonic in elevation. Furthermore, the natural length of the sella in Bourret's Horseshoe Bat coincided with a fiducial point in the relationship between sella length and beamwidth, where the effect of sella length on beamwidth saturates. This allows the prediction of the natural sella length from its acoustic properties. The long sella in this species could, hence, be an adaptation to producing a narrow beam at comparatively low frequencies.

11:20

4aAB11. An evolutionary approach to perissodactyl vocalizations. David G. Browning (Phys. Dept., Univ. of Rhode Island, 2 Lippitt Rd., Kingston, RI 02881, decibeldb@aol.com) and Peter M. Scheifele (Univ. of Cincinnati, Cincinnati, OH 45267-0379)

Approximately 65 million years ago, dinosaurs were displaced by mammals. In a then highly forested world, the perissodactyls (odd-toed ungulates) became the largest group of large mammals, consisting of an estimated 15 families. We hypothesize that they started to develop a melodic component in their tonal vocalizations in order to provide identity in this restricted visibility setting. With time, large areas shifted to grassland, favoring mammals with compound stomachs, namely, artiodactyles (sheep, cattle, etc.) and resulting in the decline of the perissodactyls to only three families (equines, tapirs, and rhinos) today. Although all the remaining perissodactyls still retain a melodic component in their vocalizations, it appears to be less developed in those that adapted to the open grassland environment, such as the plains zebra. There a simple tonal call (as the artiodactyles have, too) is sufficient to draw attention; additional information can be obtained visually. On the other hand, those that continued in a forested location such as the Sumatran Rhino have developed a more lyrical call to compensate for reduced visibility.

11:35

4aAB12. Time domain, frequency domain, and spectrogram analysis of baleen whale songs. Pranab K. Dhar and Jong-Myon Kim (Univ. of Ulsan, 680-749, Korea, jongmyon.kim@gmail.com)

Whale song is the sound to communicate, and it shows a specific pattern of regular and predictable sounds made by some species of whales. In this study, different baleen whale songs were analyzed in terms of time domain, frequency domain, and spectrogram representation. More specifically, each whale song in the species was analyzed with a sound analysis tool in terms of peak frequency (frequency with peak energy), frequency range of song, and pattern of song production. Cordell North Canyon humpback whales produce the highest peak frequency (1,348 Hz) whereas North Eastern Pacific blue whales produce the lowest peak frequency (18 Hz). The humpback whales provide wide frequency range from 600 Hz to 2.8 kHz, but Atlantic Ocean fin whales provide short frequency range from 15 to 40 Hz. Patterns of song production for the species were also analyzed. Minke and right whales generate similar repeated song. However, humpback, bowhead, blue, and fin whales generate same repeated song. These evaluation techniques can provide solutions for characterizing specific features of whale songs. [Work supported by the MKE (Ministry of Knowledge Economy), Korea, under the ITRC (Information Technology Research Center) support program supervised by the IITA (Institute of Information Technology Assessment) (IITA-2008-(C1090-0801-0039)).]

Session 4aBB

Biomedical Ultrasound/Bioresponse to Vibration: Image Enhancement and Targeted Drug and Gene Delivery

Azzdine Y. Ammi, Cochair
Oregon Health and Science Univ., Portland, OR 97239

Saurabh Datta, Cochair
Dept. of Biomedical Engineering, Univ. of Cincinnati, Cincinnati, OH 45242-0586

Invited Papers

8:00

4aBB1. Sonothrombolysis for acute coronary syndromes: Opportunities and challenges. Sanjiv Kaul (Cardiovascular Div., Oregon Health & Sci. Univ., Portland, OR 97239)

Percutanous interventions and pharmacological thrombolysis are the current options for treatment of acute coronary syndromes (ACS). The former is limited by its availability and the latter by its efficacy. Sonothrombolysis has been demonstrated to be effective in achieving tissue perfusion in the peripheral arteries as well as in cerebral arteries. Therefore, there is potential of using sonothrombolysis for the treatment of ACS. An ultrasound imaging and delivery system could overcome the issue of access while combining it with microbubbles and low dose thrombolytics could result in a high reperfusion rate. Furthermore, the direct effect of ultrasound on ishemic myocardium (release of nitric oxide and increase in myocardial blood flow despite total coronary occlusion) could be exploited to protect the myocardium until reperfusion has been achieved. The success of reperfusion could be assessed in real-time using microbubbles. In order to achieve these goals, we need to plan systematic *in vitro* and *in vivo* studies to better understand the mechanics of sonothrombolysis. We also need to develop 4D combined imaging and ultrasound delivery systems. Positive developments in this field can translate into a major impact on human health world wide.

8:20

4aBB2. Targeted microbubble technology and ultrasound-mediated gene delivery. Jonathan Lindner (Cardiovascular Div., Oregon Health and Sci. Univ., 3181 S.W. Sam Jackson Park Rd., Portland, OR 97239, lindnerj@ohsu.edu)

There is interest in harnessing the energy from ultrasound-microbubble interactions for therapeutic gain. Microbubbles (MB) can be used as vectors for ultrasound mediated gene delivery (UMGD). This process relies on coupling cDNA to MB that undergo cavitation resulting in delivery/transfection from controlled bioeffects. Proximity of gene-laden MB to the vessel wall is a critical determinant of UMGD, which may be enhanced by the use of targeted MB probes that bind to certain disease states. In this presentation, endothelial targeting strategies will be discussed together with their recent application to optimize UMGD. Flow chamber and intravital microscopy studies have been performed that demonstrate that the coupling of cDNA and MB surfaces does not interfere with binding of these agents to counterligands on the vascular endothelium. Imaging experiments have also been performed demonstrating that MB retention in tissue using an ischemia-targeting approach is possible with gene-laden MBs targeted to endothelial cell adhesion molecules. Finally, data on the relative transfection efficiency of reporter genes with UMGD using a targeted versus control MB vector approach will be discussed. The overall conclusion of the presentation is that the ability to target MB to disease-related molecules may increase both efficiency and specificity of tissue transfection.

8:40

4aBB3. Gas body destruction reduces the effective circulating dose of ultrasound contrast agent infused in rats. Douglas L. Miller, Chunyan Dou (Dept. of Radiology, Univ. of Michigan, 1301 Catherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu), and Roger C. Wiggins (Univ. of Michigan, Ann Arbor, MI 48109)

Glomerular capillary hemorrhage (GCH) in rat kidney provides a model system for assessing *in vivo* gas body efficacy in diagnostic or therapeutic applications of ultrasound. Two diagnostic ultrasound machines were utilized: one monitored the second-harmonic B mode contrast-enhancement of the left kidney and the other exposed the right kidney for GCH production. Definity contrast agent was infused at 5 μl/kg/min for 300 s during shams and 1.5-MHz intermittent exposures at 2.3-MPa peak rarefactional pressure amplitude in groups of five rats. The left kidney image brightness enhancement, indicative of circulating gas body dose, was 18.4 au (decompressed arbitrary units) in shams with no GCH in histology. Exposure of the right kidney with a normal 1-s image interval induced 68.4% GCH but reduced the left kidney enhancement to 3.3 au, which implies substantial gas body destruction. Decreased exposure with 10-s interval reduced right kidney GCH ($P' < 0.001$) but only to 30.3% while ameliorating gas body destruction with 13.1-au left kidney enhancement. The effective *in vivo* gas body dose in rats may be reduced greatly due to gas body destruction in the small animal, complicating extrapolation to similar conditions of human exposure. [Work supported by NIH grant EB00338.]

9:00

4aBB4. Combined effect of ultrasound and liposomal doxorubicin on AT2 Dunning tumor growth in rats: Preliminary results. Lucie Somaglino (Inserm U556, 151 cours Albert Thomas, 69424 Lyon cedex 03, France and Univ. de Lyon, Lyon F-69003, France, lucie.somaglino@inserm.fr), Guillaume Bouchoux, Sabrina Chesnais, Anis Amdouni, Jean-Louis Mestas (Inserm U556, 69424 Lyon cedex 03, France), Sigrid Fossheim, Esben A. Nilssen (Epitarget AS, 0307 Oslo, Norway), Jean-Yves Chapelon, and Cyril Lafon (Inserm U556, 69424 Lyon cedex 03, France)

Previous in vitro studies conducted in our group have shown the feasibility of monitoring drug release from liposomes by an inertial acoustic cavitation index. We currently report in vivo experiments utilizing the cavitation index in combined treatment of AT2 Dunning tumor grafted rats with focused ultrasound and liposomal doxorubicin. Sixty-three rats were allocated into seven groups: control, low level ultrasound treatment, high level ultrasound treatment, free doxorubicin+high level ultrasound treatment, and liposomal doxorubicin, liposomal doxorubicin+low level ultrasound treatment, and liposomal doxorubicin+high level ultrasound treatment. Based on pharmacokinetic studies, it was decided to apply ultrasound to the tumor 48 h after drug injection. An experimental setup was built to perform repeatable and rapid sonications of tumors monitored by the cavitation index. Tumor growth was assessed for a period of 35 days after tumor inoculation. Results showed that liposomal doxorubicin significantly slowed down tumor growth. However, the synergy between ultrasound and liposomal doxorubicin could not be firmly demonstrated. The lack of synergy may be due to inefficient induction of drug delivery in vivo or too high liposome dosage hiding synergistic effects. [Work funded by the Norwegian Research Council (NANOMAT programme). Epitarget AS is acknowledged for the supply of liposomes.]

9:15

4aBB5. Ultrasound-enhanced drug delivery through sclera. Robin Shah and Vesna Zderic (Dept. of Elec. and Comp. Eng., The George Washington Univ., 801 22nd St. NW, Washington, DC 20052, zderic@gwu.edu)

Achieving an increase in drug delivery through the sclera is important in the treatment of the back of the eye diseases including macular degeneration, diabetic retinopathy, etc. Our objective is to utilize therapeutic ultrasound in enhancing drug delivery through the sclera. Porcine sclera was placed in a standard diffusion cell at a normal physiological temperature of 34 °C. Solution of sodium fluorescein, a hydrophilic drug-mimicking compound, was added to donor compartment, and receiver compartment was filled with saline. The sclera was exposed to ultrasound for 5 min (intensities 1.2–1.8 W/cm^2 and frequencies 0.5 to 5 MHz). After 60 min, solution samples were taken from the receiver compartment to determine the concentration of sodium fluorescein. The sclera permeability to the drug mimicking agent *in vitro* increased 3.5 times at 0.5 MHz (*p*-value of less than 0.05), 1.7 times at 1 MHz, 3.5 times at 3.5 MHz (*p*-value of less than 0.05), and 1.5 times at 5 MHz. The average temperature of the sclera during ultrasound exposure was 42 °C. No gross changes were observed in the sclera due to ultrasound application. Future work will focus on determination of optimal ultrasound parameters for the drug delivery through the sclera.

9:30

4aBB6. Direct observation of microbubble interactions with ex vivo microvessels. Hong Chen, Andrew A. Brayman, Michael R. Bailey, and Thomas J. Matula (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105, hongchen@apl.washington.edu)

The interaction between microbubbles with tissue is poorly understood. Experimental evidence, supported by numerical simulations, suggests that bubble dynamics is highly constrained within blood vessels. To investigate this further, a high-speed microimaging system was set up to study the effects of acoustically activated microbubbles on microvessels in ex vivo rat mesentery tissues. The microbubble-perfused tissues were placed under a microscope and insonified with MHz ultrasound. A variety of interactions was observed by a high-speed camera: arterioles, venules, and capillaries were all recorded to dilate and invaginate by activated microbubbles. For small diameter microvessels, dilation and invagination were nearly symmetric, and bubble-induced rupture of the vessel was observed at high pressure. For larger microvessels, the portion of the vessel nearest the bubble coupled the strongest to the bubble dynamics, and the extent of dilation was smaller than invagination. Tissue jetting toward the bubble was recorded in many cases. The interaction of multiple bubbles inside microvessels was also observed. Bubble oscillation, vessel wall velocity, and tissue jet velocity were quantitatively measured. Invagination of vessel walls, especially tissue jetting, may be the major mechanism for tissue injury by a bubble. [Work supported by NIH 5R01EB000350.]

9:45

4aBB7. Shell buckling enhances subharmonic behavior of phospholipid coated ultrasound contrast agent microbubbles. Jeroen Sijl, Timo Rozendal, Marlies Overvelde, Valeria Garbin, Benjamin Dollet, Nico de Jong, Detlef Lohse, and Michel Versluis (Phys. of Fluids Group, Univ. of Twente, Enschede, The Netherlands)

Subharmonic behavior of coated microbubbles can greatly enhance the contrast in ultrasound imaging. The threshold driving pressure above which subharmonic oscillations are initiated can be calculated from a linearized Rayleigh-Plesset-type equation. Earlier experimental studies on a suspension of phospholipid-coated microbubbles showed a lower threshold than predicted from traditional elastic shell models. Here we present an experimental study of the subharmonic behavior of individual BR-14 microbubbles (Bracco Research) with initial radii between 1.6 and 4.8 μm. The subharmonic behavior was studied as a function of the amplitude and the frequency of the driving pressure pulse. The radial response of the microbubbles was recorded with the Brandaris ultrahigh-speed camera, while the resulting acoustic response was measured with a calibrated transducer. It is shown that the threshold pressure is minimum near a driving frequency equal to half the resonance frequency of the bubble, as expected. We found a threshold pressure as low as 10 kPa for certain bubble sizes, which can be explained by the shell buckling model proposed by [Marmottant *et al.*, JASA (2005)]. We show that the origin of subharmonic behavior is a result of the discontinuous transition within the bubble shell from the elastic state to the tensionless buckling state.

10:00—10:30 Break

10:30

4aBB8. Shell buckling increases the nonlinear dynamics of ultrasound contrast agents at low acoustic pressures. Marlies Overvelde (Phys. of Fluids, Univ. of Twente, P.O. Box 217, 7500 AE Enschede, The Netherlands, m.l.j.overvelde@utwente.nl), Benjamin Dollet, Valeria Garbin (Univ. of Twente, Enschede, The Netherlands), Nico de Jong (Experimental Echocardiography, Thoraxcenter, Erasmus MC, Rotterdam, The Netherlands), Detlef Lohse, and Michel Versluis (Univ. of Twente, Enschede, The Netherlands)

The key feature of ultrasound contrast agents in distinguishing blood pool and tissue echoes is based on the nonlinear behavior of the bubbles. Here we investigate the nonlinear properties of the shell which lead to an increased nonlinear bubble response, especially at low acoustic pressures. The microbubbles were studied in free space away from the wall using the Brandaris camera coupled to an optical tweezers setup. The microbubble spectroscopy method [Van der Meer et al., JASA, **121**, 648 (2007)] was employed to characterize BR-14 microbubbles (Bracco, Geneva). For increasing applied pressures the bubble resonance curves become asymmetrical and the frequency of maximum response decreases, up to 50% at a pressure of 25 kPa. It was found that the skewing of the nonlinear resonance curve is the origin of the so-called thresholding behavior below resonance. Traditional bubble models account for a purely elastic shell predict linear behavior, whereas the shell buckling model by Marmottant et al. [JASA, **118**, 3499

(2005)], originally developed to predict compression-only behavior, captures the asymmetry of the resonance curve in great detail. The full understanding of the nonlinear behavior at low acoustic pressures opens a wealth of new possibilities in contrast-enhanced ultrasound imaging.

10:45

4aBB9. Influence of finite wall impedance on contrast agent bubble behavior near a membrane. Todd Hay, Marlies Overvelde, Benjamin Dollet, Valeria Garbin, Nico de Jong, Detlef Lohse, and Michel Versluis (Phys. of Fluids, Univ. of Twente, 7500 AE Enschede, The Netherlands)

Experiments investigating the radial dynamics of ultrasound contrast agent (UCA) microbubbles in the vicinity of an optically transparent membrane show that bubble oscillation amplitude and frequency of peak response decrease as bubbles move closer to the membrane (Overvelde *et al.* Proc. 19th Intl. Congr. Acoust., 2007). However, treating the membrane as a rigid wall predicts that the peak oscillation amplitude should increase at small standoff distances, contrary to experimental observations. Here we present a model describing UCA bubble dynamics near a locally reacting wall having finite acoustic impedance. If finite wall impedance is included in the model, the predicted bubble behavior is in good agreement with observations. The hybrid time-frequency domain model is based on a linear frequency domain solution [Ingard, J. Acoust. Soc. Am. **23**, 329 (1951)] which has been adapted to account for weakly nonlinear bubble oscillations and UCA shell dynamics. Wall impedance parameters are derived from independent experimental measurements. Comparisons between the model and data from microbubble spectroscopy experiments will be presented. [Work supported by the ASA Hunt Fellowship and NIH DK070618.]

11:00

4aBB10. Model for bubble dynamics in liquid near an elastic tissue interface. Todd A. Hay (Phys. of Fluids, Univ. of Twente, 7500 AE Enschede, The Netherlands), Evgenia A. Zabolotskaya, Yurii A. Ilinskii, and Mark F. Hamilton (The Univ. of Texas at Austin, Austin, TX 78713-8029)

A model is under development for the weakly nonlinear dynamics of a bubble in a blood vessel surrounded by tissue with elasticity and losses. The model requires knowledge of the radiation impedance of the bubble within the vessel, which is determined by the Green's function. As a first step toward this objective, the Green's function is derived for a bubble in a liquid half-space bounded by an elastic half-space. The method used to derive the Green's function follows an approach described previously for the response of an acoustically driven object in an elastic half-space [Zabolotskaya *et al.*, J. Acoust. Soc. Am. **124**, 2514(A) (2008)]. The Green's function is expressed in terms of its angular spectrum in a plane parallel to the interface, resulting in ordinary differential equations in the coordinate normal to the interface. Boundary conditions are satisfied at the interface and in the plane containing the source to obtain solutions of the differential equations, the inverse spatial Fourier transform of which yields the desired Green's function. Extension to multiple layers of tissue is straightforward. Calculations will be presented for the radiation impedance of the bubble near the interface. [Work supported by the ASA Hunt Fellowship and NIH DK070618.]

11:15

4aBB11. Determination of maximum response radius at therapeutic pressure levels. Kelsey J. Carvell and Timothy A. Bigelow (Dept. of Elec. and Comput. Eng., Iowa State Univ., 2011 Coover Hall, Ames, IA 50010, kcarvell@iastate.edu)

High-intensity, focused ultrasound therapy is a minimally invasive therapy technique that has shown potential in many therapeutic applications, especially when coupled with the cavitation of microbubbles. The purpose of this study was to determine the effect of pressure amplitude on cavitation resonance frequency/bubble size at therapeutic field levels. Earlier work has indicated that the resonance frequency depends on pressure amplitude; however, the investigation only considered pressure amplitudes up to 1 MPa [Ultrasonics **43**, 113 (2004)]. Our study was conducted by simulating the response of bubbles using the Gilmore-Akulichev formulation to solve for the bubble response. The frequency of the sine wave varied from 1–5 MHz while the amplitude of the sine wave varied from 0.0001–13 MPa. The resonance size for a particular frequency of excitation and amplitude was determined by finding the initial bubble size that resulted in the maximum bubble expansion relative to the initial size for an air bubble in water prior to the first collapse. Preliminary simulations indicated that this metric gave the correct resonance size for small excitations. These simulations demonstrated a downshift in resonance size with increasing pressure amplitude.

11:30

4aBB12. Subharmonic response from single, polymer-shell contrast agents subjected to high-frequency excitation. Parag V. Chitnis, Jonathan Mamou, Paul Lee, and Jeffrey A. Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., 156 William St., 9th Fl., New York, NY 10038)

Three polymer-shelled agents (nominal mean diameters of 0.56, 1.1, and 3.4 μm) from POINT Biomedical were investigated to determine the optimal parameter space for generating a subharmonic response from single agents when subjected to high-frequency (above 20 MHz) excitation. A flow phantom was constructed to restrict the flow of a dilute contrast agent solution to a small volume. Two single-element transducers, with nominal center frequencies of 40 and 20 MHz, were aligned such that they were confocal and orthogonal to each other, and their mutual focus was positioned within the flow phantom. The 40 MHz transducer was used in transmit/receive mode while the 20 MHz transducer was used passively in the receive mode. The radio-frequency backscatter signals from individual contrast agents were digitized simultaneously from both transducers under a variety of pulse durations (5–20 cycles), acoustic pressure levels (1.5–6.0 MPa), and driving frequencies (35–45 MHz). Echo signals from individual contrast agents were windowed and spectra were calculated. For each contrast agent, the magnitude of the subharmonic (20–MHz) response was normalized with respect to the magnitude of the fundamental (40–MHz) backscatter. Experimental results agreed closely with theoretical calculations. [Work supported by NIH EB006372.]

11:45

4aBB13. Dissolution and stabilization of contrast microbubble by its elastic shell. Kausik Sarkar and Amit Katiyar (Mech. Eng., Univ. of Delaware, Newark, DE 19716)

We have developed a model for gas transport from encapsulated contrast microbubbles used for ultrasound imaging and drug delivery. The model explicitly accounts for permeability of gas through encapsulation and encapsulation elasticity. We use it to investigate dissolution time and stability of a lipid-coated perfluorocarbon bubble such as Definity. Realistic values of material properties such as diffusivities, permeabilities, and Ostwald coefficients of air and octafluoropropane were used to find that the lifetime of hours for such a bubble is only possible at extremely low surface tension in the absence of shell elasticity. Dissolution dynamics is investigated for effects of various parameters such as initial mole fraction of octafluoropropane, initial radius, surface tension, and shell permeability. The dissolution dynamics scales with permeability, in that when the time is nondimensioanlized with permeability, curves for different permeabilities collapse on a single curve. For an elastic shell, one obtains stable bubbles when the appropriate condition on surface tension and encapsulation elasticity is satisfied. The condition varies with the level of air saturation of the surrounding medium. For the oversaturated case, the dynamics allows multiple branches of equilibrium solutions. Stability of these branches are numerically investigated and discussed.

Session 4aID

Interdisciplinary: Workshop on Preparing Articles for the Journal of the Acoustical Society of America (JASA) and JASA Express Letters

Ning Xiang, Chair
Architecture, Rensselaer Polytechnic Inst., Troy, NY 12180

Chair's Introduction—8:55

Invited Papers

9:00

4aID1. Achieving publication excellence in the Journal of the Acoustical Society of America. Ning Xiang (Graduate Prog. in Architect. Acoust., Rensselaer Polytech. Inst., Troy, NY 12180)

Publications in refereed acoustics journals are of significant relevance for scientists and acousticians in the acoustics-related fields. The Journal of the Acoustical Society of America (JASA) encourages authors to submit papers for JASA publication. In achieving publishing excellence, the JASA regularly publishes detailed, updated guidance and instructions. As a frequent reviewer, an author of JASA papers, and an Associate Editor of the JASA, this paper will provide an outline and overview of its peer-review process of the JASA to analyze possible reasons for successful and unsuccessful publication effort. This paper also discusses how to prepare qualified manuscripts, how to avoid unnecessary delays for publications in the JASA, and how to review manuscripts for JASA.

9:25

4aID2. Realities of publishing in a journal: Why should you submit, what should you submit, and what problems might you encounter? Allan D. Pierce (Acoustical Society of America, Ste. 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502)

Attractions of a journal are its widespread availability, its archival (forever!) nature, its priority in literature searches, and its prestige. Articles should be readable to others in the field, be significant, and original. JASA is selective and imposes standards. Perceived quality is often measured by the "impact factor," which may have very little to do with the extent to which the journal fulfills the mission of the Society. Of great frustration to the editors is that a substantial fraction of authors who submit papers seem to be somewhat clueless as to what is a reasonable topic and what is a reasonable scope for an article in JASA. Frustrating also is that most good work reported at Society meetings never gets submitted. The present talk critically discusses the selection process, its peer-review aspects, and its flaws. The selection process deals with realities, such as that willing and competent reviewers are hard to find and that submitted reviews are often inane, with carefully selected highly expert associate editors who can make authoritative decisions without absolute reliance on external reviews. Suggestions are given on how to prepare a suitable manuscript and on how to cope with the vagaries of the peer-review process.

9:50

4aID3. Important aspects of editorial procedures for the information of inexperienced authors. Keith Attenborough (Dept. of Design, Development, Environment and Mater., The Open Univ., Milton Keynes MK7 6AA, UK, k.attenborough@open.ac.uk)

Several matters concerning the quality of papers and the editorial process have emerged from editorial activities for three acoustically-related journals (Applied Acoustics, Acta Acustica combined with Acustica, and the Journal of the Acoustical Society of America). Each of the journals suggests particular criteria to reviewers as the basis for their reviews. These factors are compared and discussed. Each of the journals offers various editorial decision options. These options and the uses that are made of them are discussed. Two common factors that influence editorial and reviewer judgements on submission quality include (a) the use of English and (b) the provision of a comprehensive and critical literature review. An increasingly important consideration is the availability of good reviewers in areas related to the topic of the paper. Sanitized examples illustrating some of these points are offered.

10:15—10:25 Break

10:25

4aID4. Preparing a submission to the Journal of the Acoustical Society of America on applied projects in architectural acoustics. Lily M. Wang (Architectural Engr. Prog., Peter Kiewit Inst., Univ. of Nebraska–Lincoln, 1110 S. 67th St., Omaha, NE 68182-0681)

The Journal of the Acoustical Society of America (JASA) encourages authors to submit articles for publication that are based on more applied projects, such as those commonly found in the technical area of architectural acoustics. However, very few of this type have been published in the recent past. Suggestions on how such articles should be prepared and how they may meet the "significance" criterion will be given, from the viewpoint of a current JASA associate editor in architectural acoustics. Additionally a review of such articles that have been published in JASA within the past few decades will be provided.

10:50

4aID5. Manuscript preparation—How Acta Acustica united with Acustica compares with the Journal of the Acoustical Society of America. Dick Botteldooren (Dept. of Information Technol., Ghent Univ., Belgium, Dick.Botteldooren@intec.ugent.be) and Michael Vorlaender (RWTH Aachen Universivy, Germany)

Scientific writing is essential for academic progress and technical innovation. Rules and ethics for writing are well-defined, and detailed guidelines for authors are available. Nevertheless, journals have a specific flavor which is not only related to the focus of content. Thus, for authors preparing a manuscript, the choice of journal is an important issue. Based on survey material and analyses of past publications, differences between Acta Acustica united with Acustica (AA.A) and JASA will be highlighted. Journal policy has to deal with the variety of manuscript content covering the span from fundamental science to technical applied papers. The readers expect both, and this is even more important for society journals like JASA and AA.A. Review processes that guarantee equal quality of technical and purely academic contributions, yet keep in mind their own specificity, are accordingly difficult to set up. Recent changes in handling the review of technical and applied papers in AA.A will be discussed.

11:15—11:45 Panel Discussion

THURSDAY MORNING, 21 MAY 2009 STUDIO SUITE, 9:00 TO 11:15 A.M.

Session 4aMU

Musical Acoustics: Musical Perception and Modeling

Diana Deutsch, Chair
Dept. of Psychology, Univ. of California, San Diego, La Jolla, CA 92093-0109

Contributed Papers

9:00

4aMU1. Absolute pitch among students in an American music conservatory: Association with tone language fluency. Diana Deutsch, Kevin Dooley (Dept. of Psychol., Univ. of California, San Diego, La Jolla, CA 92093), Trevor Henthorn (Univ. of California, San Diego, La Jolla, CA 92093), and Brian Head (Univ. of Southern California, Los Angeles, CA 90089)

Absolute pitch (AP), the ability to name a musical note in the absence of a reference note, is extremely rare in the United States and Europe, and its genesis is unclear. The prevalence of AP was examined among students in an American music conservatory, as a function of age of onset of musical training, ethnicity, and fluency in speaking a tone language. Taking those of East Asian ethnicity, the performance level on a test of AP was significantly higher among those who spoke a tone language very fluently than among those who spoke a tone language fairly fluently, which was in turn higher than among those who were not fluent in speaking a tone language. The performance level of this last group did not differ significantly from that of Caucasian students who spoke only intonation language. An advantage to early onset of musical training was found, but did not interact with the effect of language. Further analyses showed that the results could not be explained by country of early music education. The findings support the hypothesis that the acquisition of AP by tone language speakers involves the same process as occurs in the acquisition of a second tone language.

9:15

4aMU2. The effect of musical experience on describing sounds with everyday words. Mihir Sarkar, Cyril Lan, Joseph Diaz, and Barry Vercoe (The Media Lab., Massachusetts Inst. of Technol., 20 Ames St., Cambridge, MA 02139, mihir@media.mit.edu)

Musicians often use non-technical words such as “warm,” “sharp,” or “sweet” to describe sound quality. Commonplace experience indicates that the descriptions of a sound by a diverse group of musicians may vary, suggesting that musical background may influence one’s interpretation of a sound. A research study was carried out targeting 844 subjects of varying musical backgrounds where each subject had the chance to match various words to sound samples. Each subject was assigned to one or more musical categories (strings, woodwinds, electronic, percussion, brass) based on previous musical experience, and the results were compared across categories. Statistical measures were employed to determine if a correlation existed between musical background and survey responses. After analyzing the results from all sound-word combinations, it was determined that the musical background had no effect on the selection of words. Because of the nature of statistical hypothesis testing, the expected rate of false positives was greater than the proportion of statistically significant sound-word combinations. From the data collected in this user study, it is reasonable to suggest that the description of musical sounds is an innate skill that is not influenced by musical background and training.

9:30

4aMU3. A spectral analysis of the tuba mirum section of six live concert performances of Verdi’s Requiem as conducted by Arturo Toscanini. Robert C. Chapman (11721 W. Brandt Ave., Littleton, CO 80127)

Arturo Toscanini conducted Verdi’s Requiem 29 times with ten different orchestras between the years 1902 and 1951. Of these 29 performances, six of them were recorded between the years 1938 and 1951 with three different orchestras. An analysis of the Tuba Mirum section of this work was done to determine musical element consistency between different dates of performance and different orchestras. The Tuba Mirum section was selected because of the amount of control a conductor has to have over their musicians in order to obtain maximum musical effect. The musical elements considered in the analysis were rhythmical consistency, tempo, articulation, and dynamics. The amount of time to perform this section of music was also investigated.

9:45

4aMU4. Auditory effect of perturbing musical tones by interpolation with other musical tones. James W. Beauchamp (Dept. of Elec. & Comput. Eng. and School of Music, Univ. of Illinois at Urbana-Champaign, 1002 Eliot, Urbana, IL 61801, jwbeauch@uiuc.edu), Andrew B. Horner (Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Kowloon, Hong Kong), and Richard H. Y. So (Hong Kong Univ. of Sci. and Technolgy, Clear Water Bay, Kowloon, Hong Kong)

The effect of blending variable amounts of a secondary tone with a primary tone by means of interpolating between their corresponding harmonic amplitudes was investigated. Original tones (bassoon, clarinet, flute, horn, oboe, alto saxophone, trumpet, and violin) were normalized with respect to fundamental frequency, duration, attack and decay times, and loudness; also, harmonic frequencies were flattened. However, the basic time variations of the tones' harmonic amplitudes were preserved. Interpolation was accomplished in the frequency domain using interpolation levels between 5 and 50%. While the effect of perturbation is highly dependent on the primary /secondary instrument pair, results show that the effect of a secondary instrument is heard most easily for primary instruments horn and bassoon and least easily for primary instruments trumpet and saxophone. On the other hand, clarinet and trumpet are heard most easily as secondary instruments, whereas bassoon and violin are slow to be heard. Discrimination scores were correlated with different spectrotemporal measures of the tone spectra: spectral incoherence, normalized spectral centroid deviation, and spectral irregularity. The only significant effect found was that primary instruments with high spectral incoherence tend to mitigate against perturbation by secondary instruments. [Work supported by Research Grants Council Grants 613806 and 613508.]

10:00—10:15 Break

10:15

4aMU5. Physical modeling of the piano: An investigation into the effect of string stiffness on the hammer string interaction. Charalampos Saitis (Computational Acoust. Modelling Lab., Music Technol./CIRMMT, Schulich Sch. of Music, McGill Univ., Montreal, QC, Canada, charalampos.saitis@mail.mcgill.ca), Sarah Orr, and Maarten van Walstijn (Queen's Univ. Belfast, Belfast, UK)

The stiff string wave equation has four solutions, two of which are fast-decaying waves introduced by the string stiffness. In the case of digital waveguide modeling of piano strings these are normally neglected. Some recent reports have suggested that all four traveling waves should be considered, at least at the neighborhood of interaction points (i.e., the hammer and the boundaries). This paper investigates the effect of omitting string stiffness in the context of sound synthesis of the piano by physical modeling. A stiff, lossy string with a spatially distributed hammer force excitation is implemented using both a finite-difference time-domain scheme and a digital waveguide model. The two models are designed so as to have the exact same features but for the two stiffness-related solutions. Numerical experiments are employed to study the contact force and string velocity signals for different initial hammer velocity values. The results generally confirm that the two fast-decaying waves have only a marginal effect on the overall string motion. However, small audible differences result for bass strings struck with high initial hammer velocities.

10:30

4aMU6. A real-time music synthesis engine for physical modeling of plucked string instruments. Huynh V. Luong (School of Comp. Eng. and Info. Tech., Univ. of Ulsan, Ulsan 680-749, Korea), Sangjin Cho, Jong-Myon Kim, and Uipil Chong (Univ. of Ulsan, Korea)

Music synthesis continues to offer huge potential possibilities for the creation of new musical instruments. One of the promising music synthesis techniques is physical modeling which produces output sounds that resemble much more closely their physical counterparts since it offers the potential of more intuitive control. However, this results in tremendous computational and I/O requirements, prohibiting real-time use in composition and live performance. To meet the performance requirements, we introduce a parallel processing engine. Whereas commercial digital signal processors such as TI TMS320C6x families use silicon area for large multiported register files, large caches, and deeply pipelined functional units, our parallel processing engine contain many more simple processing elements (PEs) for the same silicon area. As a result, our engine often employs thousands of PEs while possibly distributing and colocating PEs with the data I/O to minimize storage and data communication requirements. In this paper, we implemented physical modeling of a representative plucked Korean string instrument, called Gayageum, which has 12 silk strings. Our engine achieved 12-notes music synthesis in real-time at 44.1 kHz sampling rate for the physical modeling algorithms. This is in contrast to TI TMS320C6x, which achieves only single-note music synthesis. [Work supported by KOSEF, R01-2008-000-20493-0.]

10:45

4aMU7. Acoustical properties of pure sound piano wire. David Ripplinger, Brian Anderson, Tim Leishman, and William Strong (Dept. of Phys., Brigham Young Univ., Provo, UT 84602)

Pure sound piano wire is a stainless-steel wire that has only recently entered the U.S. market. Because of its different composition, which makes the wire more malleable, it should have a considerably lower amount of inharmonicity compared to regular steel wire. Measurements were conducted on several pianos with regular piano wire (Rosslau and Mapes) and pure sound in order to assess their differences in inharmonicity and tonal qualities. The pure sound wire produced measurably less inharmonicity than regular wire on the same kind of piano. However, this difference is much smaller compared to the difference between small and large pianos. This presentation will explain the methods of measurement and analysis, as well as simulations that were implemented in order to analyze the effect that inharmonicity has on the temperament, octave stretching, and interval patterns. [The Brigham Young University Department of Physics is acknowledged for the funding it provided for this research.]

11:00

4aMU8. Identification of impact sounds by professional percussionists. Ching-Ju Liu (Dept. of Communicative Disord., Univ. of Wisconsin, 1975 Willow Dr. Madison, WI 53706, chingjuliu@wisc.edu) and Robert Lutfi (Univ. of Wisconsin, Madison, WI 53706)

The present study was undertaken to determine the best of the ear's ability to identify the sounds of impact produced by the simplest of resonant objects (bars, plates, and membranes). We hypothesized that best performance would be achieved by professional percussionists who have had many years experience striking such objects to achieve desired nuances in sound. Five percussionists and five nonpercussionist musicians were recruited as participants from the University of Wisconsin School of Music. Additionally, 10 nonmusicians were recruited from the University at large. In a standard two-interval, forced-choice procedure, with and without feedback, listeners were asked to judge the impact sound corresponding to: (1) the greater force of impact; (2) the harder of two mallets; (3) the denser of the two objects struck; and (4) the point of mallet contact closest to the center of the object. Sounds were synthesized according to the equations of motion derived from a simple physical model which has been used in past studies [cf. Lutfi *et al.*, JASA 118, 393–404 (2005)]. Generally, the results provide little support for the hypothesis that professional percussionists are more adept than the rest of us at judging the properties of impact sounds. [Research supported by NIDCD.]

Session 4aNS

Noise, Architectural Acoustics and ASA Committee on Standards: Hospital Noise and Health Care Facilities

Erica E. Ryherd, Chair
Dept. of Mechanical Engineering, Georgia Inst. of Technology, Atlanta, GA 30332-0405

Invited Papers

9:00

4aNS1. The new comprehensive acoustical criteria for healthcare facilities. David M. Sykes and Gregory Tocci (Remington Partners, LLC, 23 Buckingham St., Cambridge, MA 02138)

New comprehensive acoustical guidelines for healthcare facilities were drafted for the American Institute of Architects and the American Hospital Association in 2005–6 by the 500-member TC-AA.NS.SC (Speech Privacy & Healthcare Acoustics) which is chaired by Gregory Tocci, David Sykes, and William Cavanaugh. Following two rounds of public review, these guidelines were approved for use by the Green Guide for Healthcare in 2007 and in early 2009 they were issued as the sole "Reference Standard" for two new LEED "Indoor Environmental Quality" credits. In early 2010, following two additional rounds of public review, the same guidelines will be released by the Facility Guidelines Institute and the American Society of Healthcare Architects (a division of the American Hospital Association) in the authoritative, code-level document, "FGI/ASHE Guidelines for the Design and Construction of Hospitals and Healthcare Facilities." Practitioners and researchers need to be aware of these criteria when designing acoustical solutions for healthcare facilities. This paper will present an overview of the guidelines. The speaker is co-chairman of the technical committee responsible for drafting the criteria and of ANSI S12 WG44, which is currently developing an additional new healthcare acoustics standard.

9:20

4aNS2. Comparing the sound environments in two critical care settings. Selen Okcu (GaTech, College of Architecture, Atlanta, GA 30332-0155), Erica Ryherd, and Craig Zimring (GaTech, Atlanta, GA 30332-0155)

Critical care nurses perform crucial tasks in complicated sound environments. The existence of many different noise sources (i.e., staff conversation, medical alarms, etc.) with different sound qualities cause nurses to experience constantly changing acoustic conditions while providing care for critically ill. Some of those acoustical qualities in critical care settings can negatively affect nurse well-being and task performance. In different critical care settings, the acoustic qualities can vary. Some architectural qualities of those settings can be an indicator of those variations. In this study, we documented the detailed objective and subjective sound environment of two critical care settings with different architectural layouts. The comparative analyses are used to understand the differences between two acoustic settings and the relationship between subjective and objective sound environments. Perceived qualities of the physical work environments are examined to explore the association between architectural features and acoustic qualities.

9:40

4aNS3. Evaluating the intensive care unit soundscape. Timothy Y. Hsu, Erica Ryherd (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, gth776e@mail.gatech.edu), and Kerstin Persson Waye (Gothenburg Univ., 405 30 Gothenburg, Sweden)

The intermittent sounds in hospital wards may induce arousal among patients leading to responses such as annoyance, sleep disturbance, and cardio-vascular reactions. The sound environment as a whole may also affect the efficiency and general health among the staff. A series of studies are being conducted by the authors to evaluate the modern hospital soundscape including occupant response. Collaborations between engineering and medicine are being utilized to assess the soundscape from both a quantitative and qualitative standpoint. This talk will focus on soundscape evaluations of intensive care units. This includes a pilot study performed in a medical-surgical intensive care unit (ICU) of a Swedish hospital. Patients were monitored for 24 hours during their stay in the ICU and both acoustic and physiological data were simultaneously recorded. Additionally, the staff wore dosimeters and completed perception questionnaires. The methodology and analyses of these detailed acoustic measurements and preliminary subjective results will be discussed. [Work supported by ASA and Swedish FAS.]

10:00—10:15 Break

10:15

4aNS4. Mechanical equipment and ambulances. Strange bedfellows of cardiovascular services. Chad N. Himmel and Jack B. Evans (JEAcoust., 1705 W Koenig Ln., Austin, TX 78756, himmel@jeacoustics.com)

A case study will be presented regarding vibration and noise control design for an existing heart hospital. Facility expansion within a crowded site necessitates sandwiching a suite of new cardiac catheterization laboratories between mechanical equipment rooms above and the hospital's main ambulance driveway below. The facility expansion also includes cardiac computed tomography, magnetic reso-

nance imaging, and nuclear imaging, plus cardiovascular care, treatment, and administrative support spaces. Procedure and support spaces can be disturbed by mechanical equipment noise and vibration and noise from ambulance siren. Cardiac procedure instruments and imaging can be disturbed by low frequency noise and vibration from building mechanical systems, ambulance traffic, and building occupants. Design criteria for permissible building vibration and continuous background noise will be presented along with relevant sources and bases. Measurement evaluation results for design will be contrasted with criteria. Measures incorporated in design to mitigate vibration and noise on catheterization laboratories will be discussed. Measures considered in design include structural floor "detuning" concepts, stiffened supports for catheterization laboratory C-arms, mechanical equipment vibration isolation mounts, and floating concrete floor systems for mechanical rooms. If available, postconstruction spectral analysis measurements of building noise and vibration will be presented.

10:35

4aNS5. Predicting structure-borne noise for new construction adjacent to medical facilities. Richard A. Carman (Wilson, Ihrig & Assoc., Inc., 5776 Broadway, Oakland, CA 94618) and Gary M. Glickman (Wilson, Ihrig & Assoc., Inc., New York, NY 10004)

When new building construction is conducted in close proximity to existing medical or medical research facilities, structure-borne noise due to construction activities generating vibration can cause impacts to the occupants of the existing building. Theoretical modeling of structure-borne noise is very complicated and can be extremely time consuming. The feasibility of the two primary candidates (i.e., FEA and SEA) for modeling this phenomenon is questionable, at least at this point, for a number of reasons that are discussed. It is possible that the time will come when a theoretical method can be used reliably to predict structure-borne noise transmission. In the meantime, using an empirical approach seems to be the most viable. A case study is presented of a facility for medical research involving mice. Preconstruction projections of structure-borne noise and vibration are compared to those measured with an extensive monitoring program employed during construction.

10:55

4aNS6. Vibration sensitivity of optical microscopes in the healthcare setting. Hal Amick and Michael Gendreau (Colin Gordon & Assoc., 150 North Hill Dr., Ste. 15, Brisbane, CA 94005, Hal.Amick@colingordon.com)

The paper examines vibration criteria for benchtop and articulated floor-supported optical microscopes, comparing published specifications with data measured in hospitals for diagnostic purposes. A case study of vibrations that degraded orthopedic microsurgery is of particular interest. Revisions and enhancements of published criteria are proposed.

Contributed Paper

11:15

4aNS7. Soundscape analysis of a neonatal intensive care unit (NICU) facility. Gary Siebein (School of Architecture, Univ. of Florida, Gainesville, FL 32611-5702, gsiebein@siebeinacoustic.com), Reece Skelton, Victoria McCloud, Robert Lilkendey, and Hyun Paek (Siebein Assoc., Inc., Gainesville, FL 32607)

A soundscape study was made of an existing NICU facility in a major urban hospital to document the nature and magnitude of sounds experienced by newborn infants in this environment. Focus group discussions were held with hospital administrators, NICU staff, design team members, and families of patients to determine the types of sounds that are heard in normal operation of the NICU. Long-term average sound level measurements of general sounds in the NICU were made for several work shifts. Short-term acoustical measurements of specific acoustic events were also made to obtain octave band sound level data for each of the activities, medical equipment sounds, and building infrastructure sounds that comprise the soundscape of the unit. Audio and video recordings of the specific acoustical events that comprise the soundscape of the NICU were also made. The acoustical measurements and soundscape analysis are used to evaluate proposed acoustical design criteria for NICU facilities compared to nonacoustic health risks reported in the literature.

Session 4aPA

Physical Acoustics, Underwater Acoustics, and Engineering Acoustics: A Half-Century with the Parametric Acoustic Array I

Kenneth G. Foote, Cochair
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Murray S. Korman, Cochair
Physics Dept., U. S. Naval Academy, Annapolis, MD 21402

Chair's Introduction—7:55

Invited Papers

8:00

4aPA1. Early years of the parametric array—An anecdotal history. David T. Blackstock (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029 and M.E. Dept., UT Austin, Austin, TX 78712-0292)

This review covers the period from 1960 to about the mid-1970s. Westervelt presented his theory of the parametric array at the June 1960 Acoustical Society Meeting, held at Brown Univ. In the same session, Bellin and Beyer reported experimental confirmation, primarily for the array in water. Despite the tantalizing properties of the parametric array—narrow low-frequency beam produced with a small source, absence of sidelobes, and potentially broadband operation—the acoustical community at first paid little attention. Berktay woke us up with a series of papers in the mid-to-late 1960s on possible applications. Interest in underwater parametric arrays then exploded, including attempts to make parametric receiving arrays practical. Definitive experimental confirmation of the parametric array in air was finally reported in 1973. However, practical applications of the airborne array did not appear until many years later.

8:20

4aPA2. Parametric acoustic arrays: A Bergen view. Halvor Hobæk (Dept. of Phys., Univ. of Bergen, Allegt. 55, N-5007 Bergen, Norway)

At the University of Bergen (UoB), Norway, research activity in physical acoustics started in the mid-1960s with investigations on the parametric acoustic array (PAA). The newly appointed professor in applied mathematics, Sigve Tjøtta, had some years earlier been at Brown University and was inspired by the concept at a fundamental level, but also wanted experimental confirmation. No previous acoustical activity existed at UoB. The PAA project was started as a master project at Department of Physics, where the main activity was in nuclear, high-energy, and ionospheric physics. Bellin and Beyer's experiment served as a model. The results provided new information on the axial and directional properties of the difference frequency wave field. Inspired by this, theoretical modeling continued along with further measurements. Other nonlinear effects like acoustic streaming (boundary layer, density gradient) were also investigated. In 1975, a project together with SIMRAD and Norwegian Technical University resulted in a bottom penetrating PAA, later commercialized as "TOPAS." Numerical modeling based on the KZK equation resulted in the "Bergen Code," still in use for computing nonlinear acoustic propagation problems. In later years activity at UoB has expanded to encompass linear physical acoustics of various sorts, occasionally using PAA as a tool.

8:40

4aPA3. Parametric acoustic array development at the U.S. Navy's New London, Connecticut laboratory. David G. Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com), Mark B. Moffett (731 Annaquatucket Rd., North Kingstown, RI 02852), and William L. Konrad (54 Laurel Hill Dr., Niantic, CT 06357)

A brief history of the development of parametric acoustic sources at the U.S. Navy Underwater Sound Laboratory (USNUSL), and its successor, the Naval Underwater Systems Center is presented. Inspired by Robert Mellen, the Parametric Sonar Group was formed to explore the practical implications of Westervelt's idea for underwater acoustics. Spanning more than two decades, this research pursued various potential applications exploiting the unique characteristics of parametric sources, including echo-ranging in reverberant environments, communications involving voice, data, music, and video signals, target-strength measurements, bottom bathymetry, high-resolution sub-bottom profiling, and measurements of scattering and reflection from surfaces. Parametric source design procedures were developed, and guidelines were established for diagnosing and avoiding undesirable effects, such as difference-frequency instability and spurious nonlinearities, such as direct radiation from the projector, cavitation, and receiver nonlinearity. Theoretical models extended the Westervelt theory to realistic primary fields. For planar projector arrays, parametric source design curves, valid for levels in the primary far field, enabled designers to predict performance without resorting to computer calculations. To obtain source-levels and beam-widths within the primary near field, where the secondary signal is generated, a computer program, called CONVOL5 in its present configuration, can be used to obtain more detailed performance predictions.

9:00

4aPA4. Research on parametric arrays in Russia: Historical perspective. Lev A. Ostrovsky (Zel Technologies/Univ. of Colorado, Boulder, CO 80305)

This presentation is concerned with the Russian part of the history of parametric arrays (PAs). After the pioneering works of P. Westervelt in the United States and V. Zverev and A. Kalachev in Russia and a decade of relative isolation, Soviet and Russian researches became involved internationally in the work on the PAs and, more generally, of nonlinear acoustic beams. Several Russian institutions have been involved, including Moscow University, Andreev Institute of Acoustics in Moscow, Radiophysics Research Institute and Institute of Applied Physics in Gorky (now Nizhny Novgorod), Gorky University, Pacific Oceanological Institute in Vladivostok, and some others. Selected Russian contributions of different years are to be outlined in this talk, including: (1) The theory of nonlinear acoustic beams, e.g., the Khokhlov-Zabolotskaya- Kuznetsov equation and other models of nonlinear acoustic beams in application to the PA; (2) the limitation of the PA efficiency by shock wave formation; (3) possible enhancement of the PA radiation by bubble layers; (4) experiments in laboratory and in the ocean; and (5) some new developments such as generation of low-frequency shear waves by ultrasound beams with biophysical applications.

9:20

4aPA5. Reflections on the early days of parametric array research and development. Thomas Muir (Natl. Ctr. for Phys. Acoust., Univ. of Mississippi, One Coliseum Dr. University, MS 38677)

Following Westervelt's classic paper (1960/1963), there was a period of laboratory exploration, involving small-scale experiments and theoretical modeling, with Orhan Berktay's early work anticipating the significance of several practical applications. Robert Mellen organized the First International Symposium on Nonlinear Acoustics (ISNA) in 1968, which the preset author was privileged to attend. Research interest and momentum began to build, expanding to field and seagoing experiments, as well as computational modeling. Some of the seminal milestones and achievements of this period are identified and illustrated. These include theoretical work, which enabled new experiments, and vice versa, facilitated by research tool engineering. A number of classic problems were studied, including parametric transmission and the effects of finite amplitude attenuation. Applications to air, water, and sediment research evolved, including propagation and backscattering measurements, subbottom profiling, and modal effects and communication in shallow water. Parametric reception became a hot topic, along with its intricate signal processing. Focusing of finite amplitude beams was developed, involving the use of lenses, also utilized for steering and scanning parametric beams. The bandwidth capability of parametric interaction was recognized, as were parametric transients for echo-ranging. Imaging applications emerged for both harmonic and difference frequency radiations, and a spin-off to biomedical ultrasonics was forged. By the mid-1980s, a dozen ISNA symposia had been held and the field had blossomed.

9:40

4aPA6. Audio parametric arrays in air. Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713-8029)

Following Westervelt's discovery and theoretical description of the parametric acoustic array reported at the 1960 meeting of the ASA at Brown Univ., and the measurements in both water and air at ultrasonic frequencies, confirming Westervelt's predictions qualitatively, reported by Bellin and Beyer at that same meeting, it was not until 1973 that Bennett and Blackstock reported convincing quantitative comparisons of theory and experiment for a parametric array in air. While their experiments produced difference frequencies in the audio range, not until the late 1990s did application of the parametric array to directional audio transmission in air achieve any commercial success. This presentation will briefly review the history of developments leading up to modern-day audio parametric arrays. Challenges associated with transducer performance, signal processing, and acoustical measurements specific to audio parametric arrays in air will be described. Features of several audio parametric arrays currently on the market will be discussed. Finally, results will be presented from an ongoing collaboration between the author's group at Univ. of Texas at Austin and Professor Pierre Khuri-Yakub's group at Stanford Univ. to create audio parametric arrays in air using capacitive micromachined ultrasonic transducers to transmit the ultrasonic primary beams.

10:00—10:20 Break

10:20

4aPA7. High-powered parametric acoustic array in air. Robert W. Haupt (MIT Lincoln Lab., Active Optical Systems Group, 244 Wood St., Lexington, MA 02420, haupt@LL.mit.edu)

MIT Lincoln Laboratory has developed a prototype high-powered parametric acoustic array (HPPAA) for standoff acoustic excitation in several applications. Parametric arrays offer a highly directional, narrow beam mechanism to deliver sound in air to desired targets typically within a 100 m range. However, a difficult challenge arises in generating sufficient sound power at difference-frequencies below 1000 Hz at a range which can be critical for many target types. An important objective of the HPPAA design maximizes the difference-frequency pressure amplitude at range by maximizing the end-fire array length established by the PAA. The design optimizes the trade-offs between the three characteristic lengths that control the resultant end-fire array length including the pump or carrier wave attenuation, PAA aperture, and acoustic saturation of air. In field demonstrations, the HPPAA generated a carrier wave pressure power approximately 155 dB re-20-microPa one meter from the PAA face while generating a 300 Hz difference-frequency SPL of 90 dB re-20-microPa 8 meters from the PAA. These sound pressure levels at a few hundred hertz may enable safe standoff excitation and detection of buried landmines and may be useful in standoff nondestructive testing (NDT) damage detection and imaging of structures.

10:40

4aPA8. The development of a 75 kilohertz phase steered active parametric sonar system for subseabed target detection. Paul Lepper and Bryan Woodward (Dept. Electric and Elec. Eng., Loughborough Univ., Loughborough, LE11 3TU UK, p.a.lepper@lboro .ac.uk)

Parametric sonar systems offer a number of potential advantages over conventional sonar's that has encouraged investigation and implementation of nonlinear acoustic effects into sonar systems by numerous groups since Westervelt's discovery reported in 1956. The work represented here describes the development of an active parametric sonar system utilizing a 75 kHz primary frequency signal with measured secondary frequency generation from 1–22 kHz using a 2-D 729 element square array arranged into 13 staves of 54 elements each. The array, dedicated computer controlled signal synthesis and power amplifier systems were developed and tested during trials on the Fisheries Research Services trails site on Loch Duich, Scotland. Primary and secondary frequency source levels of 245 dB *re* 1 μPa-m and 195 dB *re* 1 μPa-m at 7 kHz were measured with −3 dB full angle beam-widths of 3 and 2.5°–7° for frequencies 3–11 kHz, respectively. Individual control of the thirteen stave (channels) allowed in-pulse scanning and beam steering across a 36° sector including demonstration of real-time dynamic stabilization in one plane. Results will be presented for implementation of the system into a towed body sonar used during sediment classifications trials off the coast of France and for target detection tasks in Scotland.

11:00

4aPA9. The parametric array in Berea sandstone: definitive experiments. Pierre-Yves Le Bas, James A. TenCate, Robert A. Guyer, and Paul A. Johnson (Geophys., MS D443, Los Alamos Natl. Lab., Los Alamos, NM 87545, pylb@lanl.gov)

Previous measurements of the characteristics of the parametric array in sandstone by Johnson and Shankland [J. Geophys. Res. **94**, 17729–17734 (1989)] were difficult to perform and only qualitative. Scanning laser vibrometers (Polytec) now make parametric array measurements in rock easier. However, hysteresis and memory effects play a strong role in the dynamic behavior of rocks and have the potential to mess up the creation of the parametric array in the medium. Thus, an experimental study was performed to find out just how well the "classical" theory of nonlinear acoustics works for a granular solid, a sandstone. An array of alternating PZT disk transducers was epoxied to a large block of Berea sandstone (1.2, 0.4, 0.4 m), every other one broadcasting with separate frequency generators and amps. Primary frequencies were around 100 kHz; difference frequencies of 10 to 20 kHz were observed. Two-D beampattern scans were taken and the results compared with calculations based on classical theory. The agreement is surprisingly good with a beta of around 200 and a Q (inverse attenuation) of 50.

11:20

4aPA10. Laboratory applications of truncated parametric arrays. Victor F. Humphrey (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, SO17 1BJ, United Kingdom, vh@isvr.soton.ac.uk.)

Parametric arrays have characteristics that make them ideally suited for use as versatile sources in the laboratory for performing a range of measurements and acoustic studies. Such measurements can exploit the wideband character of the source to provide detailed acoustic response characteristics of systems as a function of frequency, while the short pulses and narrow beamwidths enable multiple reflections and unwanted echoes to be resolved or minimized. However, the need to make measurements in the nearfield of the array means that truncation of the nonlinear interaction region with an acoustic low-pass filter is advisable in order to generate a region free of secondary-sources and simplify the interpretation of the results. This truncation also avoids the possible influence of hydrophone nonlinearity. The effects of truncation on the array characteristics are illustrated using both model and experimental results. A number of examples of the use of arrays in the laboratory are given, including their use to study acoustic scattering from structures and diffraction from baffles, and measurement of reflection and transmission properties of materials. These demonstrate how the characteristics of parametric arrays can be exploited to make detailed, accurate measurements of relevance to underwater acoustics under laboratory conditions.

11:40

4aPA11. Parametric projectors protecting marine mammals from vessel collisions. Edmund R. Gerstein (Leviathan Legacy Inc., 1318 SW 14th St., Boca Raton, FL 33486, gerstein2@aol.com), Laura A. Gerstein (Leviathan Legacy Inc., Boca Raton, FL 33486), and Steven E. Forsythe (U.S. Naval Undersea Warfare Ctr., Newport, RI)

Marine mammals are vulnerable to ship collisions. Measurements of controlled ship passages through vertical hydrophone arrays demonstrate a confluence of propagation factors and near surface effects that obscure the sounds of approaching vessels which then pose serious detection challenges for marine mammals. Joe Blue, who first identified these challenges, later conceived of a parametric method to mitigate them. A highly directional, dual-frequency parametric array has been developed to reduce collision risks by selectively alerting only those animals in the direct path of approaching vessels. The system projector is comprised of multiple elements, band-centered to transmit a high carrier frequency along with a lower side band signal. A single-side band modulation and phase-shift method are employed. The non-linearity of water then demodulates the mixed high frequency carrier into a lower frequency waveform audible to both manatees and whales. The bow mounted array projects a narrow beam directly ahead of vessels, and "fills in" acoustical shadows to alert marine mammals of approaching danger. Controlled field tests of the manatee device in Florida's NASA wildlife refuge are proving effective. Real-world deployments on select Navy and DOD vessels are planned this year and sea tests of a larger whale system will start next year. [Funded by U.S. Department of Defense, Navy Legacy Resource Management Program.]

Session 4aPP

Psychological and Physiological Acoustics: Auditory Spatial Perception

G. Christopher Stecker, Chair
Dept. of Speech and Hearing Science, Univ. of Washington, Seattle, WA 98105

Contributed Papers

9:00

4aPP1. Revealing and quantifying temporal aspects of envelopes of high-frequency stimuli that lead to enhanced processing of interaural temporal disparities. Leslie R. Bernstein and Constantine Trahiotis (Depts. of Neurosci. and Surgery (Otolaryngol.), Univ. of Connecticut Health Ctr., Farmington, CT 06030)

This presentation concerns evaluating which aspects of envelopes of high-frequency waveforms foster efficient discrimination of interaural temporal disparities (ITDs). The experiments were designed to assess, quantitatively, the explanatory capability of metrics including: (1) the normalized fourth moment of the envelope (*Y*), (2) the normalized interaural correlation of the envelope, and (3) the dead time between peaks or maxima of the envelope. An ITD-discrimination study was conducted employing raised-sine stimuli centered at 4 kHz [John *et al.*, Ear and Hearing **23**, 106–117 (2002)]. The parameters of the stimuli that could be varied using such stimuli include: (1) the exponent of the raised sine, (2) the frequency of modulation, (3) the depth of modulation, (4) the degree of phase-scrambling applied to the spectral components of the envelope, and (5) the degree of phase-scrambling applied to the spectral components of the waveform. This set of parameters permitted us to construct an ensemble of stimuli that pit one metric against another in a manner that reveals the relative salience of particular temporal aspects of the envelope. Results will be discussed both in terms of the characteristics of the external, physical stimuli versus the stimuli as processed by the auditory periphery. [Research supported by research Grant Nos. NIH DC-04147 and DC-04073 from the National Institute on Deafness and Other Communication Disorders, National Institutes of Health.]

9:15

4aPP2. Sensitivity to interaural level differences in high-rate high-frequency click trains: A functional magnetic resonance imaging assessment. G. Christopher Stecker and Susan A. McLaughlin (Dept. of Speech & Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105, cstecker@u.washington.edu)

While the auditory cortex (AC) is known to be necessary for accurate sound localization, the nature of spatial representation in AC remains poorly understood. Past animal studies have quantified sensitivity to auditory spatial cues [e.g., interaural level difference (ILD)] across subregions of AC. In this study, functional magnetic resonance imaging (fMRI) was used to assess ILD sensitivity across the surface of human AC. Normal-hearing listeners were presented with narrowband Gabor click trains (4000 Hz center frequency, 3 ms interclick interval) that varied parametrically in ILD over a range of +/− 30 dB. Simultaneously, whole-head fMRI images were acquired using a sparse imaging paradigm at 3 Tesla (32-slice EPI, 3.0×3 .0×4.5 mm resolution, TR=12 s). During imaging, listeners detected rare target changes in ICI. Hemodynamic responses of compact AC subregions in individual subjects were assessed quantitatively using statistical pattern recognition with two goals: (1) to identify contiguous AC regions that share common patterns of sensitivity to ILD, and (2) to quantify, in information-theoretic terms, the ability of those responses to differentiate stimuli differing in ILD.

9:30

4aPP3. Precedence effect in children and adults: Effects of heard and unheard echoes on localization accuracy. Ruth Litovsky, Shelly Godar, and Phillip Wesolek (Waisman Ctr., Univ. of Wisconsin, 1500 Highland Ave., Madison, WI 53705, litovsky@waisman.wisc.edu)

The precedence effect is thought to facilitate sound localization in reverberant environments through localization dominance. This is demonstrated at short lead-lag delays, when a single fused auditory image is heard whose perceived location is dominated by the leading source. We studied localization dominance in children ages 4–5 and in adults. Stimuli were brief noise bursts, with lead-lag delays ranging from 1–100 ms; the lead location varied along the azimuth (−60, −40, −20, 20, 40, 60 deg) and the lag was fixed at 0 deg. Subjects indicated the perceived location(s) of each heard source. Localization accuracy was computed for the lead and lag at each delay. On a separate task, with delays varying randomly, subjects indicated whether they heard one or two auditory events, and echo threshold was obtained. Results show that localization accuracy for the lead declined as delays increased, suggesting that as the lag became more audible its interference in the localization process increased. For most subjects, the interference was predictable from independent measures of fusion echo thresholds. The impact of the lag, a single simulated echo, on localization accuracy, was greater for children than adults. These data suggest that the precedence effect is less effective at facilitating sound localization in young children than in adults. [Work supported by NIDCD Grant R01DC008957.]

9:45

4aPP4. Real-virtual equivalent auditory localization with head motion. Griffin D. Romigh and Douglas S. Brungart (Air Force Res. Lab., Wright-Patterson Air Force Base, OH 45433)

Several researchers have shown that individualized head related transfer functions (HRTFs) can be used to produce virtual sounds that are equivalent in terms of localization to free field sounds. Thus far, however, these results have only been shown in studies that have required listeners to keep their heads stationary during the playback of the virtual sounds. In this study, we investigated the performance limits of a virtual auditory display system that incorporated individualized HRTFs but allowed free head motion during the localization process. One key aspect of the system is a high-speed HRTF measurement process that allowed a full set of individualized HRTFs to be measured in less than 4 min. This made it possible to make an HRTF recording and complete a localization task using the resulting HRTFs within the same 30-min experimental session. The results show that equivalent free-field and virtual localization performance was achieved when the virtual sounds were generated in the same session using specially-designed open-ear headphones that did not need to be removed during the headphone equalization process. This indicates that equivalent real-virtual closed-loop localization is possible even with the truncated, interpolated, minimum-phase HRTF filters that are required for practical, real-world virtual audio systems.

10:00—10:30 Break

10:30

4aPP5. Head-related transfer function enhancement for improved vertical-polar localization. Douglas S. Brungart, Griffin D. Romigh, and Brian D. Simpson (AFRL/RHCB, Wright-Patterson Air Force Base, 2610 Seventh St., OH 45433)

Under ideal laboratory conditions, individualized head-related transfer functions (HRTFs) can produce virtual sound localization performance approaching the level achieved with real sound sources in the free field. However, in real-world applications of virtual audio, practical issues such as fit-refit variability in the headphone response and nonindividualized HRTFs generally lead to much worse localization performance, particularly in the up-down and front-back dimensions. Here we present a new technique that "enhances" the localizability of a virtual sound source by increasing the spectral contrast of the acoustic features that are relevant for spatial perception within a set of locations with nearly identical binaural cues (i.e., a "cone-of-confusion"). Validation experiments show that this enhancement technique can improve localization accuracy across a broad range of conditions, with as much as a 33% reduction in vertical-polar localization error for nonindividualized HRTFs measured on a KEMAR manikin; a 25% reduction in vertical-polar error for nonindividualized HRTFs measured on other human listeners; and a 33% reduction in vertical-polar error for individualized HRTFs presented under nonideal laboratory conditions (i.e., with headphone fit-refit variability). These results suggest that the proposed technique could provide benefits across a wide range of real-world virtual audio display applications. [Work sponsored by AFOSR.]

10:45

4aPP6. Azimuth dependency in auditory perception of speaker's facing angle. Hiroaki Kato, Hironori Takemoto, Ryouichi Nishimura, and Parham Mokhtari (NICT/ATR, 2-2-2 Hikaridai, Seika-cho, Soraku-gun, Kyoto 619-0288, Japan, kato@atr.jp)

In pursuit of an ultimately realistic human-to-human telecommunication technology, the ability to auditorily perceive the facing direction of a human speaker was measured. A male speaker sat on a pivot chair in an anechoic chamber and spoke a short sentence (about 5 s) while facing either of eight azimuth angles (0=listener's direction, 45, 90, 135, 180, 225, 270, or 315 deg). These angles were set solely by turning the pivot chair. Blindfolded listeners heard the spoken sentence at a distance of either 1.2 or 2.4 m from the speaker and were asked to indicate the speaker's facing angle. The average of errors between listeners' responses and actual speaker's facing angles was 23.5 deg. Further analyses showed a significant effect of the speaker's facing azimuth on the accuracy of listeners' responses. Listeners made remarkably small errors (6.6 deg on average) when the speaker faced the listener's direction (azimuth = 0 deg). The second smallest errors were observed for the back angle (azimuth = 180 deg). Few front-back confusions were observed. This dependency of listeners' accuracy was extensively discussed from the interest of effective acoustic cues obtained by a precise computer simulation as well as an ecological perspective.

11:00

4aPP7. Strength and cardivascular fitness predict time-to-arrival perception of looming sounds. John G. Neuhoff, Katherine L. Long, and Rebecca C. Worthington (Dept. of Psych., The College of Wooster, Wooster, OH, jneuhoff@wooster.edu)

Perceiving rapidly approaching sounding objects can be critical for survival. In studies of "auditory looming" perception, listeners consistently perceive sound sources as closer than they actually are, resulting in an underestimation of arrival time (Neuhoff, Planisek, and Seifritz, 2009; Rosenblum, Carello, and Pastore, 1987). This effect has been argued to provide an evolutionary advantage by allowing more time to prepare for the source. However, critical to this argument is the timely engagement of motor behaviors. Here, we tested the hypothesis that listeners with lower levels of physical fitness would have a larger anticipatory bias in perceived auditory arrival time, and thus a larger margin of safety in response to looming sounds. Listeners judged the arrival time of a three-dimensional looming sound. Physical fitness was measured using recovering heart rate after exercise and grip strength. Results show that the anticipatory bias in perceiving looming sounds is negatively correlated with physical fitness ($r=-0.41$). Those least prepared physically to interact with a looming sound source have a greater perceptual margin of safety. The findings are consistent with an evolutionary explanation of the anticipatory bias for looming sounds and provide evidence for fitness-based perception-action links between the auditory and motor systems.

11:15

4aPP8. How visual cues help us understand speech in a complex environment. Lingqiang Kong and Barbara Shinn-Cunningham (Cognit. and Neural Systems, 677 Beacon St., Boston, MA 02215)

At a cocktail party, visual cues may help a listener by showing them *where* or *when* to direct attention, *what* acoustic modulations a target utterance contains, and/or *what* articulatory gestures produce the target. Here, we investigated target speech intelligibility while varying the visual cues available in a complex, confusing auditory scene. In all cases, subjects listened for a target utterance in the presence of multiple masker utterances with similar grammatical structure spoken by the same talker. The timing and direction of the target (and maskers) varied randomly, increasing the uncertainty about where and when to focus auditory attention. The number of correctly reported target key words measured performance. Performance tended to improve as the amount of visual information increased, particularly when masker phrases came from the direction of the target. Performance was generally similar whether listeners saw full videos of the target talker from the correct direction or only a static image of the talker at the right time in the correct direction. However, temporal information about where and when the target occurred improved performance over knowing only target location. Results suggest that in these scenes, visual cues aid target understanding by indicating where and roughly when to direct attention.

11:30

4aPP9. Stimulus continuity is not necessary for the salience of dynamic sound localization cues. Ewan A. Macpherson (School of Commun. Sci. and Disord. and Natl. Ctr. for Audiol., Univ. of Western ON, 1201 Western Rd., London, ON, Canada, N6G 1H1, ewan.macpherson@nca.uwo.ca)

Correspondence between head rotation and resulting changes in interaural difference cues provides information about sound source location. We assessed whether source continuity or merely relative displacement is necessary for use of this dynamic localization cue. Low-frequency (0.5–1 kHz) noise-band targets, not correctly localizable in the absence of head motion or for motion duration <50 ms [Macpherson and Kerr, APCAM (2008)], were presented while the listener performed a practiced head rotation at 50 deg/s. The stimuli were either continuous (a single burst gated on and off as the head entered and exited a variable-width spatial window) or discrete (two 20 ms endpoint bursts, triggered as the head entered and exited the window). Human listeners reported the apparent location of the stimulus by orienting with their heads subsequent to the initial head rotation. The minimum head movement angle (MHMA) necessary to resolve front/rear ambiguity was measured for each stimulus type. Similar MHMAs of 5–10 deg were measured for continuous and discrete stimuli, suggesting that endpoint "snapshots'' providing only displacement information are sufficient for use of dynamic localization cues. That parallels the finding that stimulus continuity does not improve detection of source motion [Chandler and Grantham, J. Acoust. Soc. Am. 106, 1956–1968 (1992)]. [Work supported by NSF and NIH/NIDCD.]

11:45

4aPP10. Localizing a speech target in a multitalker mixture. Norbert Kopčo (Dept. of Cybern. and AI, Tech. Univ. of Košice, Letná 9, Košice, Slovakia, kopco@bu.edu), Virginia Best, and Simon Carlile (Univ. of Sydney, Sydney, Australia)

Despite the importance of spatial hearing in everyday listening, little is known about the accuracy of sound localization in a complex mixture of sounds. Here we measured, for the frontal audio-visual horizon, how accu-

rately listeners could localize a female-voice target amidst four spatially distributed male-voice interferers in a moderately reverberant room. To examine whether listeners can make use of *a priori* knowledge about the configuration of the scene, we compared performance when the interferer locations were fixed (in one of five known patterns) to when the locations varied from trial to trial. The presence of interferers disrupted target localization, even after accounting for reduced target detectability. Randomizing the interferer locations had a moderate influence, degrading performance in some configurations but improving it in others. All effects were magnified when the target-to-interferer intensity ratio was reduced. The results confirm that spatial perception is disrupted by interfering sounds, and that this disruption is modified to some extent by listeners' expectations about the spatial arrangement of the scene. [Work supported by HFSP, NIH, VEGA, ARC and a Univ. of Sydney Postdoctoral Fellowship.]

THURSDAY MORNING, 21 MAY 2009 FORUM SUITE, 9:00 TO 10:00 A.M.

Session 4aSAa

Structural Acoustics and Vibration: Distinguished Lecture

Sean F. Wu, Chair
Dept. of Mechanical Engineering, Wayne State Univ., Detroit, MI 48202

Chair's Introduction—9:00

Invited Paper

9:05

4aSAa1. A residual-potential boundary for time-domain problems in computational acoustics. Thomas L. Geers (Dept. of Mech. Eng., Univ. of Colorado, Boulder, CO 80309, geers@spot.colorado.edu)

For many years, researchers have sought theoretically exact time-domain computational boundaries that are temporally, spatially, and geometrically local. Unfortunately, spatial locality and geometric locality cannot be simultaneously achieved for exact boundaries. *Absorbing* boundaries offer spatial locality but not geometric locality, whereas *impedance* boundaries offer the latter but not the former. Following a brief discussion of these topics, a new, theoretically exact impedance boundary is introduced that is based on modal *residual potentials* for the spherical geometry. The new boundary produces a set of first-order, uncoupled ODEs for *nodal* boundary responses and a set of low-order, uncoupled stepping formulas for *modal* boundary responses. The two sets are coupled through nodal-modal transformation based on the orthogonal surface functions for the spherical boundary. Numerical results generated with the boundary are presented for selected benchmark problems. Finally, extension of the method to other separable geometries for the wave equation is discussed.

THURSDAY MORNING, 21 MAY 2009 FORUM SUITE, 10:15 TO 11:55 A.M.

Session 4aSAb

Structural Acoustics and Vibration: Vibro-Acoustic Diagnosis and Prognosis of Complex Structures I

Wen Li, Chair
Dept. of Mechanical Engineering, Wayne State Univ., Detroit, MI 48202

Invited Paper

10:15

4aSAb1. Experimental validation of vibroacoustic reconstruction of a rectangular plate with different boundary conditions. Logesh Kumar Natarajan and Sean Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202)

This paper provides an experimental validation of vibroacoustic responses of a rectangular plate reconstructed using Helmholtz equation least squares (HELS) method. Experimental studies were conducted on baffled rectangular plates of different aspect ratios with free as well as clamped boundary conditions under random excitation. The radiated acoustic pressures were measured using a planar array of microphones at a very close distance to the plate surface and taken as input for the HELS codes. The normal surface velocity distributions were then reconstructed, and the results were compared against the benchmark data obtained using a laser vibrometer. The structural modes of the plate with different boundary conditions were compared with those obtained by experimental modal analysis. Good agreements were obtained for both clamped and free boundary conditions.

10:40

4aSAb2. Dynamic modeling of complex structures in a broad frequency range. Hongan Xu and Wen L. Li (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202)

A general method, the so-called finite substructurE method (FSEM), is presented for the dynamic analysis of complex structural systems. In this method, a complex structure is considered as a collection of a finite number of basic structural components such as beams, plates, and shells. Instead of seeking a numerical solution at a number of discrete or grid points, the current displacement solution is sought, over each component, as a continuous field in the form of Fourier series expansions. Thus, the number of degrees of freedom, which now represent the Fourier expansion coefficients, can be substantially reduced in comparison with a grid-based solution for the same spatial resolution. Mathematically, the resulting system tends to be better conditioned than those in the finite element methods as the number of DOF's increases with frequency. The robustness of this model for high frequency applications can be further improved by incorporating statistical processes into the modeling method to properly reflect the means or account for the uncertainties of certain input variables. The proposed substructure method is considerably different from the existing substructure techniques in that no modal data are required for any component. Numerical examples are presented to demonstrate the reliability of this method.

11:05

4aSAb3. Vibro-acoustic coupling of a rectangular cavity backed by a flexible panel with general boundary conditions. Jingtao Du, Zhigang Liu, Tiejun Yang (College of Power and Energy Eng., Harbin Eng. Univ., Harbin, 150001, People's Republic China), and Wen Li (Wayne State Univ., Detroit, MI 48202)

Vibro-acoustic coupling system composed of a rectangular cavity and a flexible panel is widely studied in the backgrounds of panel vibration analysis and active noise control in enclosed sound field either by sound or structural actuators, corresponding to ANC and ASAC, respectively. In the most of the current investigation, the boundary conditions of the flexible structure are limited to simple case, namely simply supported and/or clamped. As an important structural parameter, boundary condition has a great effect on the coupling characteristics of such vibro-acoustic system, and a good understanding on this phenomenon will be helpful to the system design as well as active noise control. In this presentation, the analytical model of a rectangular cavity backed by a flexible panel with general boundary conditions is developed. Two sets of improved 2-D and 3-D cosine series with supplementary terms are constructed to respectively describe the displacement and pressure distributions in the boundary structure and 3-D acoustic cavity. The aim of introduction of the supplementary terms is to overcome the potential discontinuity encountered along the structural boundary and the fluid–solid coupling interface respectively for the both fields mentioned above. Numerical calculations are carried out to show the effectiveness of the current method and to study the effect of the structure boundary on the coupling characteristics of the panel-cavity system.

11:30

4aSAb4. Semi-active shock control technique for two-stage vibration isolation system based on vibration absorption. Wanyou Li, Zhigang Liu, Xueguang Liu, and Xixia Chen (College of Power and Energy Eng., Harbin Eng. Univ., Harbin, 150001, People's Republic of China)

Active control of shock is more difficult due to its instantaneous nature. In this presentation, the use of vibration absorption is proposed to decrease the harm from shock. Simulation and experimental studies are carried out on a two-stage vibration isolation system. The dynamic model of the two-stage vibration isolation system with the semi-active vibration absorber installed is developed. The effect of various parameters of semi-active absorber is analyzed through numerical simulation. The correctness is also validated by the experimental studies. The results show that the optimal installed position is on the upper stage mass, and a set of the optimal absorber parameters exists to make the anti-shock effectiveness achieve best. Such best effectiveness is not affected by the shock amplitude, however, by the continuance time and the excitation waveform. A semi-active vibration reduction and shock resisting control system including electromagnetism semi-active vibration absorber, DSP controller, and constant current source; etc is also designed. The experimental studies on the control effectiveness and reaction time of such system are subsequently performed. It is demonstrated that such a system can work in two working conditions, namely, vibration reduction and shock resisting, these two statutes can transiently switch, and control the secondary shock response effectively.

Session 4aSCa

Speech Communication: Vowel Inherent Spectral Change

Geoffrey Stewart Morrison, Cochair
School of Language Studies, Australian National Univ., Canberra, ACT 0200, Australia

Peter F. Assman, Cochair
School of Behavioral and Brain Science, Univ. of Texas at Dallas, Richardson, TX 75083-0688

Chair's Introduction—8:00

Invited Papers

8:05

4aSCa1. Static and dynamic approaches to understanding vowel perception. James M. Hillenbrand (Dept. of Speech Pathol. & Audiol., Western Michigan Univ., Kalamazoo, MI 49008, james.hillenbrand@wmich.edu)

The purpose of this paper is to provide a broad overview of work leading up to the current view that vowel inherent spectral change (VISC) plays a significant role in the recognition of vowel identity. Although seldom if ever explicitly stated, the view that implicitly guided vowel perception research for many years assumed that nearly all of the information needed to specify vowel identity was to be found in a cross section of the vowel spectrum sampled at a reasonably steady portion of the vowel. There is now a considerable body of evidence showing that VISC plays an important role in the recognition of vowel identity. Evidence comes from: (1) measurement studies showing that many nominally monophthongal English vowels show significant spectral change throughout the course of the vowel; (2) statistical pattern recognition studies showing that vowel categories are separated with far greater accuracy by models that take spectral change into account; and (3) perceptual experiments showing that vowel steady-states are neither necessary nor sufficient for conveying vowel identity. In spite of this evidence, a static view of vowel identity continues to be implicitly assumed in many studies of vowel quality. [Work supported by NIH.]

8:25

4aSCa2. Spectral change in the front vowels of North American English. Terrance M. Nearey (Dept. of Linguist., Univ. of Alberta, Edmonton, AB T6G 2E7, Canada)

Assmann and Nearey [J. Acoust. Soc. Am. **80**, 1297–1308 (1986)] coined the term "vowel-inherent spectral change" (VISC) to refer to change in spectral properties inherent to the phonetic specification of vowels. Although VISC includes the relatively large formant movements associated with acknowledged diphthongs, it was explicitly extended to include reliable (but possibly more subtle) spectral change associated with vowel categories of North American English typically designated as monphthongs. This paper reviews statistical evidence of VISC in the formant patterns in front vowels of /hVd/ syllables in three regional dialects of English: Dallas, TX [Assmann and Katz, J. Acoust. Soc. Am. **108**, 1856–1866 (2000)], Western, MI [Hillenbrand *et al.*, J. Acoust. Soc. Am. **97**, 3099–3111 (1995)] and Northern, AB (new data). Results suggest that VISC patterns may be useful characteristics for assessing dialect differences. Evidence is presented for the importance of VISC in vowel perception, including recent evidence from our laboratories regarding perception by second language learners. A progress report is provided on research into how VISC is best characterized parametrically and which temporal regions of a vocoid may be most effective in summarizing VISC patterns in varying consonantal contexts.

8:45

4aSCa3. Formant-frequency trajectories as acoustic correlates to speech perception. Michael Kiefte, Tara Collins (School of Human Commun. Disord., Dalhousie Univ., 5599 Fenwick St., Halifax, NS B3H 1R2 Canada), Christian Stilp, and Keith R. Kluender (Univ. Wisconsin, Madison, WI 53706)

Formant trajectories are excellent vowel discriminants; within vowel, they are nearly constant across speaker size, age, and sex, and across consonantal contexts. However, this model assumes that formant peaks are perceptually important and that human listeners track formant-frequency changes across time. Speech-recognition applications have avoided formant frequencies due to the difficulty of reliable formant tracking. In addition, it is not actually known whether human listeners do indeed follow formants perceptually across time. This paper presents results from several studies that examine the relationship between changing formant frequencies and perception. Alternative perceptual representations of vowels, such as global spectral shape, are precluded by evidence that individual formant amplitudes are largely ignored in vowel perception. In addition, where other spectral properties appear to have a perceptual effect, it is because stimuli have used formants that do not change. When formants are changing, perceptual effects of spectral shape properties disappear. In terms of human formant tracking, perceptual extrapolation of a formant sweep is mostly dependent on peak frequency and not other properties related to spectral shape. This demonstrates that listeners do indeed follow formant-frequency changes as auditory objects. Further research on formant frequency perception will be described.

9:05

4aSCa4. Vowel-inherent spectral change enhances adaptive dispersion. Keith R. Kluender, Christian E. Stilp, Timothy T. Rogers (Dept. of Psych., Univ. of Wisconsin, 1202 W. Johnson St., Madison, WI 53706, krkluend@wisc.edu), and Michael Kiefte (Dalhousie Univ., Halifax, NS Canada, B3H 1R2)

Despite wide diversity among particular vowel sounds used across the world's languages, there are profound systematicities across languages. Whether sets of three, five, seven, or more vowel sounds are used, vowels that comprise these sets have substantial commonality across languages. Using static measures of vowel spectra, Lindblom and colleagues have demonstrated principles of adaptive dispersion through which the compositions of vowel inventories can be predicted on the basis of maximizing perceptual distinctiveness among the vowels within a language. Here, we address whether introduction of vowel-inherent spectral change is consistent with principles of optimizing perceptual distinctiveness between vowels. We find that vowel-inherent formant trajectories generally serve to further disperse vowel sounds across time. Trajectories of formants for vowel sounds that are relatively close in static measures (formant center frequencies: beginning, center, end) tend to be relatively distinct as measured by angles in F_1, F_2, F_3 coordinates. In a complementary fashion, vowels that share similar trajectories have relatively distinct static characteristics. This perceptual efficacy of vowel-inherent spectral change maintains across multiple place-of-articulation contexts. Across the vowel space and across consonantal contexts, vowel-inherent spectral change serves to increase adaptive dispersion and enhance perceptual distinctiveness. [Work supported by NIDCD.]

9:25

4aSCa5. Vowel-inherent spectral change and the second-language learner. Catherine L. Rogers and Merete M. Glasbrenner (Dept. of Comm. Sci. and Dis., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620)

Relatively few studies have directly examined the use of vowel-inherent spectral change by second-language learners, perhaps because it represents one of the subtler cues to vowel identity. Nevertheless, understanding non-native listeners' perception of this cue and its integration with other cues to vowel identity can be regarded as a method of investigating the mastery of cues to vowel perception that is needed for native-like perception. While several studies of second-language speech perception have demonstrated differences in cue use by second-language learners, our investigation of the use of dynamic spectral cues and temporal cues by native and non-native listeners suggests that relatively early learners of English as a second language do not appear to use the dynamic spectral cue differently from native English speakers when other cues are preserved. Instead, early learners' vowel perception appears to be less robust to removal of multiple cues. This apparent difference in even early learners' ability to use cues to vowel identity independently of one another or to change listening strategy when one cue is degraded may explain a portion of the increased challenge that even relatively early learners of a second language appear to experience in understanding speech in noisy environments. [Work supported by NIH.]

9:45

4aSCa6. Vowel inherent spectral change in forensic voice comparison. Geoffrey Stewart Morrison (School of Lang. Studies, Australian Natl. Univ., Canberra, ACT 0200, Australia, geoff.morrison@anu.edu.au)

Two-parameter models of vowel inherent spectral change, such as dual-target or target-plus-slope models, have been found to be adequate for vowel-phoneme identification. More sophisticated curve-fitting models do not appear to outperform such two-parameter models. This suggests that if only simple cues such as initial and final formant values are necessary for signaling phoneme identity, then speakers may have considerable freedom in the path taken between the initial and final formant values. If the constraints on formant trajectories are relatively lax with respect to vowel-phoneme identity, then with respect to speaker identity there may be considerable information contained in the details of formant trajectories. Differences in physiology and idiosyncracies in the use of motor commands may mean that different individuals consistently produce different formant trajectories between the beginning and end of the same vowel phoneme. If within-speaker variance is substantially smaller than between-speaker variance, then formant trajectories may be exploited for forensic voice comparison. This paper reviews a number of forensic-voice-comparison studies (including studies conducted using the likelihood-ratio framework) which have extracted information relevant to speaker identity from formant trajectories. For the purposes of forensic voice comparison, models using parametric curves are found to outperform simple two-parameter models.

Session 4aSCb

Speech Communication: Vowel Perception and Production (Poster Session)

Peter F. Assman, Cochair
School of Behavioral and Brain Science, Univ. of Texas at Dallas, Richardson, TX 75083-0688

Geoffrey Stewart Morrison, Cochair
School of Language Studies, Australian National Univ., Canberra, ACT 0200, Australia

Contributed Papers

All posters will be on display from 10:10 a.m. to 11:30 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 10:10 a.m. to 10:50 a.m. and contributors of even-numbered papers will be at their posters from 10:50 a.m. to 11:30 a.m.

4aSCb1. Developmental study of vowel-inherent spectral change. Peter F. Assmann (Univ. of Texas at Dallas, Richardson, TX 75083), Terrance M. Nearey (Univ. of Alberta, Edmonton, AB, T6G 2E7, Canada), and Sneha V. Bharadwaj (Univ. of Texas at Dallas, Richardson, TX 75083)

Children's speech differs from adult speech in several important ways. First, children have smaller larynges and supra-laryngeal vocal tracts than adults, with the result that their formants and fundamental frequencies are higher. Second, the temporal and spectral properties of children's speech are inherently more variable, a consequence of developmental changes in motor control. Both of these sources of variability raise interesting questions for theories of talker normalization and vowel specification. In the present study we compare the pattern of time-varying spectral change in vowels from a database of vowel recordings from adults and children ranging in age from 5 through 18 years from the Dallas, Texas region. Preliminary findings indicate systematic age-related differences in the average frequencies of the formants, but the pattern of vowel-inherent spectral change is well preserved across the age span investigated.

4aSCb2. Cross-generational differences in dynamic formant patterns in vowels. Robert Allen Fox and Ewa Jacewicz (Speech Percept. and Acoust. Labs., Speech and Hearing Sci. The Ohio State Univ., Columbus, OH 43210-1002, fox.2@osu.edu)

As the position of a vowel changes within the vowel space across generations of speakers, so does its dynamic formant pattern. This study examines variation in the dynamic patterns of vowel formants across two age groups: children (8–12 years) and older adults from their grandparents' generation (51–65 years). The cross-generational changes in vowels /ɪ, ɛ, æ/ were examined for each of the three regional variants of American English spoken in Southeastern Wisconsin (affected by the Northern Cities Shift), Western North Carolina (affected by the Southern Vowel Shift) and Central Ohio (not considered to be affected currently by any vowel shift). The following vowels in children's productions were monophthongized as compared to those of their grandparents' generation: Wisconsin /ɪ, ɛ/ (but not /æ/), North Carolina /ɛ, æ/ (but not /ɪ/), and Ohio /ɪ, ɛ, æ/. In addition to the reduced formant movement, some of the children's vowels had a different direction of formant change. These cross-generational changes were assessed in a set of measures including formant trajectory length, spectral rate of change and angle of formant change. The measures were calculated from formant frequencies extracted at points in the vowels corresponding to 20, 35, 50, 65, and 80% of the vowel's duration. [Work supported by NIH.]

4aSCb3. Generational and dialectal effects on children's vowel identification. Ewa Jacewicz and Robert Allen Fox (Speech Percept. and Acoust. Labs., Speech and Hearing Sci. The Ohio State Univ., Columbus, OH 43210-1002, jacewicz.1@osu.edu)

This study examines vowel identification by 8–13 years old children who grew up in either Southeastern Wisconsin (whose regional variant is affected by the Northern Cities Shift) or Western North Carolina (affected by the Southern Vowel Shift). In the first identification task, the children responded to words edited from sentences which elicited both stressed and unstressed vowel exemplars. This speech material was produced by multiple talkers representing two generations (children and older adults who represent their grandparents' generation). In the second identification task, the children were presented with citation-form tokens produced by three generations of talkers (children, their possible parents, and their possible greatgrandparents). Both within- and across-dialect vowel identification was examined. The cross-generational results showed that some vowels were identified more accurately when spoken by children, some when spoken by adults and for others there were no cross-generational differences. The cross-dialectal results indicated generally more accurate identifications of vowels produced by talkers from the same dialect region as the listeners. For selected vowels, there were significant interactions between dialect and generation. As a whole, the study shows children's sensitivity to cross-generational vowel changes and the attunement to their own dialect. [Work supported by NIH.]

4aSCb4. Vowel detection and vowel identification in long-term speech-shaped noise. Kathleen O'Brien, Ashley Woodall, and Chang Liu (Dept. of Commun. Sci. and Disord., 1 University Station A1100, The Univ. of Texas at Austin, Austin, TX 78712)

Psychometric functions of vowel detection and vowel identification were measured in long-term speech-shaped noise (LTSSN) for normal-hearing listeners. A four-interval forced-choice procedure was used to examine the accuracy of vowel detection in LTSSN with speech level presented from -10 to $+5$ dB sensational level relative to vowel detection thresholds obtained from Liu and Eddins' study [J. Acoust. Soc. Am. **123**, 4539–4546 (2008)]. The accuracy of vowel detection was significantly influenced by vowel category and sensational level. The threshold of vowel detection for each vowel and each listener was defined as the speech level at which 70.7% accuracy of vowel detection was reached. Vowel identification was then measured in LTSSN with vowel levels presented from 0 to 12 dB sensational level relative to individual thresholds of vowel detection, using a close-set 12-choice procedure. Results suggest that vowel identification was significantly affected by vowel category and sensational level. Altogether, the results of vowel detection and vowel identification indicate that, given the same signal-to-noise ratio, vowels are not equally audible and identifiable. Moreover, given the same sensational levels, vowels do not have the same identifiability, possibly due to the fact that some vowels have dominant confusing vowels while others do not.

4aSCb5. Auditory representation of vowel quality. Andrew B. Wallace and Sheila E. Blumstein (Dept. of Cognit. and Linguistic Sci., Brown Univ., Box 1978, Providence, RI 02909, andrew_wallace@brown.edu.)

It is generally assumed that the early stages of speech perception involve the extraction of some kind of generalized auditory patterns or properties from the peripheral input. The auditory representation that results is of considerable interest, since it serves as the input to higher-level, speech specific processes of phonetic perception. The current research examines this auditory representation using a priming paradigm, in which perception of vowel targets is facilitated when the targets are preceded by acoustically matched nonspeech stimuli. By manipulating the acoustic parameters of these nonspeech "prime" tones, it is possible to determine the role of these parameters in the auditory stages of vowel processing. Previous results [Wallace and Blumstein, J. Acoust. Soc. Am. **119**, 3245 (2006)] suggest a short window of analysis of no more than 25 ms. In Experiment 1, frequency of nonspeech primes was varied, with results suggesting broad frequency tuning. In Experiment 2, primes matched to both F1 and F2 of the target vowels were found to elicit a greater priming effect than would be predicted by summing the response to separately presented F1 and F2 primes, suggesting that the auditory representation of vowels encodes combinations of formant frequencies.

4aSCb6. Cross-dialect differences in vowel identification. Amy T. Neel and Cai S. Ewing (Dept. of Speech & Hearing Sci., Univ. of New Mexico, Albuquerque, NM 87131, atneel@unm.edu)

In this study, identification of vowels produced by Michigan speakers by listeners of two dialects, Michigan (Inland North) and New Mexico (Western) was examined. New Mexico listeners identified sets of ten vowels (i, ɪ, e, ɛ, æ, ɑ, o, ʌ, ʊ, u) from 30 speakers in the Hillenbrand *et al.* (1995) database. Their results were compared to the results from the Michigan listeners in the original study. Preliminary results from ten New Mexico listeners show that overall identification rates across the 30 speakers did not differ significantly between dialects. However, NM and MI listeners differed on identification rates for individual speakers and particular vowels. They agreed on the worst speaker but differed by as much as 11 percentage points for others. NM listeners performed worse on /ɛ/ and /ɑ/ but slightly better on /ʌ/ than MI listeners. To determine the nature of cross-dialect differences in perception, acoustic data such as vowel space area, formant dynamics, and duration characteristics will be examined, and the relation of acoustic data to identification scores will be presented.

4aSCb7. Effects of dialect on vowel acoustics and intelligibility. Austin L. Oder (Dept. of Speech-Lang.-Hearing, Univ. of Kansas, Dole Ctr., 1000 Sunnyside Ave., Rm. 3001, Lawrence, KS 66045, aoder@ku.edu), Sarah Hargus Ferguson (Dept. of Speech-Lang.-Hearing: Sci. and Disord., Univ. of Kansas, Lawrence, KS 66045), and Cynthia G. Clopper (Dept. of Linguist., Ohio State Univ., Columbus, OH 43210)

A great deal of recent research has focused on the phonetic variation of American English vowels from different dialects. This body of research continues to grow as vowels periodically and unconsciously undergo formant movements that become characteristic of certain dialects (e.g., the Northern Cities Chain Shift and the Southern Vowel Shift). Two experiments using the Nationwide Speech Corpus [Clopper and Pisoni, Speech Communication, **48**, 633–644 (2006)] are exploring whether the Midland dialect is more closely related acoustically and perceptually to the Southern or to the Mid-Atlantic dialect. Experiment 1 consists of acoustic analyses of 11 English vowels from each of the three dialects. In Experiment 2, 11 vowels in /hVd/ format recorded from speakers of the three dialects are being presented to speakers of a Midland dialect for identification. This study will thus further our understanding of acoustic and perceptual differences between the most marked dialects (Mid-Atlantic and Southern) and the least marked dialect (Midland). [Work supported by a University of Kansas Honors Program Undergraduate Research Award.]

4aSCb8. Spectral analysis of the vowels of the Peking dialect. Wai-Sum Lee (Dept. of Chinese, Translation and Linguist., City Univ. of Hong Kong, 83 Tat Chee Ave., Kowloon, Hong Kong)

The paper presents the spectral characteristics of the dorsal and apical vowels in open syllables of the Peking dialect. Results show that (i) of the dorsal category the high and low vowels are monophthongs, whereas the mid-back vowels and rhotacized ɚ are diphthongal; (ii) the two apical vowels are similar in F1 and F2, but differ considerably in F3, with a lower F3 for the retroflexed than the plain one; (iii) the extensive overlaps between the vowel ellipses for the unrounded mid-back vowel and plain apical vowel in the F1F2 plane and between those for the rhotacized ɚ and retroflexed apical vowel indicate that the difference in perceptual quality between the paired vowels may be attributed to the dynamic property of the formant frequencies; (iv) as expected, the average F-values for the vowels are higher for females than males, however, the patterns of positions of the vowel ellipses in the F1F2 plane between the two genders are similar; and (v) the scaling between female and male F-values is nonuniform across vowel categories and across formant frequencies. The formant data will also be discussed in connection with the quality descriptions of the vowels in the past studies of the Peking dialect.

4aSCb9. Voice identification accuracy using multivariate vowel formant analysis. Al Yonovitz and Elvan Moss (Dept. of Commun. Sci. and Disord., Univ. of Montana, Missoula, MT 59812, al.yonovitz@umontana.edu)

Accurate and automated voice or speaker identification has been a major goal for those involved in forensic issues. In addition, voice and speaker identification has many applications in security and business. Numerous previous efforts to derive features for speaker classification have failed to achieve a sufficiently low error rate. In this study a linear discriminant analysis was independently performed for vowel formants (F1, F2, F3) for each of ten vowels. The standardized canonical discriminant coefficients (SCDC) weighted each of the formants within a database of voices. These SCDC values were then linearly combined to form a single scalar for each vowel. An analysis that considered the multivariate data vector composed of the ten vowel scalars was then used for classification. This algorithm was tested by comparing single voice exemplars to other voices in the database. Accuracy was extremely high using vowel formant values for classification.

4aSCb10. A duration-dependent account of coarticulation for hyper- and hypoarticulation. Harvey M. Sussman (Dept. of Linguist. and Commun. Sci. and Disord., Univ. of Texas, 1 University Station, Austin, TX 78712, sussman@mail.utexas.edu), Bjorn Lindblom (Stockholm Univ., Stockholm, S10691, Sweden), and Augustine Agwuele (Texas State Univ., San Marcos, TX 78666)

Hyperarticulated and hypoarticulated speech are accompanied by spectral expansion and spectral reduction of vowel nuclei, respectively. These shifts in F1/F2 vowel space directly affect degree of anticipatory coarticulation in consonant + vowel sequences apart from traditional vowel context effects. Examining the opposite conditions of emphatic stress [Lindblom *et al.*, J. Acoust. Soc. Am. **121**, 3802–3813 (2007)] and faster speaking rates (Agwuele *et al.*, Phonetica, in press) it was shown that coarticulatory effects could be documented independently of the expected vowel expansion /reduction effects. A modified locus equation regression metric was used in both studies to isolate alterations in F2 transition onsets due to prosodic and speech rate conditions apart from vowel space shifts per se. The current study provides a unified empirical and theoretical account for the opposite coarticulatory effects by providing duration data as the common variable tying both studies together. Articulatory and acoustic simulations of deeper (emphasis) and shallower (increased tempo) closures are provided to explain the shifts of observed F2 onsets relative to predicted F2 onsets due simply to expanded/reduced vowel space.

4aSCb11. Variability of vowel productions within and between days. Shannon L. M. Heald and Howard C. Nusbaum (Dept. of Psych., Univ. of Chicago, 5848 S. Univ. Ave., Chicago, IL 60637, smbowdre@uchicago.edu)

Theories of speech production and speech perception assume that phonetic categories can be characterized by stable properties. For example, the notion of stable vowel targets is used to organize articulation, and such targets could serve as the acoustic basis for recognizing vowel categories. However, there is substantial variation in the acoustic patterns of phonetic categories. The lack of invariance between acoustic properties and the phonetic categories of speech has posed a theoretical problem for understanding human speech perception. Although most theories vary in how acoustic-phonetic variability is approached, theories treat such variability as inherent in speech production or statistically regular or noise. In order to begin to understand acoustic-phonetic variability, we examined the naturally occurring variability in speech production over time. Participants' productions of seven different vowels ([IH], [EE], [EH], [AW], [AH], [AE], [UW]) were recorded in nine sessions: at three specific times in the course of each testing day (9 a.m., 3 p.m., 9 p.m.), every other day, over the course of 5 days.

Formant frequencies and variability of formant frequencies were analyzed to assess how vowel categories change over time. We will discuss the results and consider the theoretical implications of these results.

4aSCb12. Acoustic vowel space in pre-/r/ contexts: Shetland and American English. Peter Sundkvist (Dept. of English, Dalarna Univ., SE-79188 Falun, Sweden, psn@du.se)

The acoustic space available to vowel systems in pre-/r/ contexts commonly differs from that of other phonetic contexts. In English this space has gradually shrunk, which relates to changes in the phonetic nature of /r/, having gone from more consonantal articulations (tap, trill) to approximant and often complex articulations that have a stronger effect on the production and perception of adjacent vowels (e.g., "bunched" and retroflex /r/). This paper contains an acoustic study of pre-/r/ vowel systems in Shetland and American English. F1–F3 values were obtained from steady state vowel portions from words spoken in isolation. The acoustic vowel spaces and the positions of contrastive items within these are compared and discussed in relation to the phonetic nature of /r/ and its effects on vowel formants. In contrast to American English, Shetland maintains a trill or tap as the principal realization of /r/, and the full range of vowel contrasts found elsewhere is supported before /r/. As Shetland is one of the most conservative English dialects in this respect, the study may also offer some insight into the acoustic characteristics of English pre-/r/ vowel systems of earlier periods and the pressures affecting such systems.

4aSCb13. Relative contribution of jaw and tongue to the vowel height dimension in American English. D. H. Whalen, Aude Noiray, Khalil Iskarous, and Leandro Bolanos (Haskins Labs, 300 George St., St. 900, New Haven, CT 06511, whalen@yale.edu)

Vowels are typically described according to three articulatory dimensions: height, frontness, and rounding. Other researchers propose a role for the jaw in the height dimension. In the present study, we measured the relative contribution of the tongue and jaw for vocalic height distinction in American English vowels. Tongue and jaw motions were collected in six adult speakers for six vowels from all portions of the vowel space, embedded in [hVd] sequences. Tongue shape was captured via HOCUS (Haskins Optically Corrected Ultrasound System) which combines digital ultrasound imaging at 127 Hz with optical three-dimensional tracking of infrared emitting diodes (IREDs) positioned on the speaker's head and probe for subsequent decoupling of tongue motion from the jaw motion. Results showed a consistent role of the tongue in creating the vocalic constriction for the six subjects investigated, though with some idiosyncratic strategies. Jaw contribution was more sizable for vowel pairs traditionally contrasted by height than for tense/lax pairs, confirming previous findings from x-ray studies. One new result is the dominance of tongue contribution over the jaw's, even for vowels distinguished by height. [Work supported by NIH grant DC-02717.]

THURSDAY MORNING, 21 MAY 2009 — PAVILION EAST, 8:00 TO 11:00 A.M.

Session 4aSP

Signal Processing in Acoustics: Pattern Recognition in Acoustic Signal Processing I

Grace A. Clark, Chair

Electronics Engineering, Lawrence Livermore National Lab., Livermore, CA 94550

Chair's Introduction—8:00

Invited Papers

8:05

4aSP1. Tutorial: Pattern recognition in acoustic signal processing. Mark Hasegawa-Johnson (Dept. of ECE, and Beckman Inst., Univ. of Illinois, 405 N Mathews, Urbana, IL 61801)

This tutorial presents a framework for understanding and comparing applications of pattern recognition in acoustic signal processing. Representative examples will be delimited by two binary features: (1) regression versus classification (inferred variables are continuous versus discrete); (2) instantaneous versus dynamic (inference algorithms consider only an instantaneous observation vector versus inference algorithms integrate observations with knowledge of system dynamics). (1. Regression) problems include imaging and sound source tracking using a device with unknown properties and inverse problems, e.g., articulatory estimation from speech audio. (2. Classification) problems include, e.g., classification of animal and human vocalizations and nonspeech audio events. Instantaneous classification and regression are performed using a universal approximator (neural network, Gaussian mixture, classification, and regression tree), regularized, if necessary, to reduce generalization error (resulting in a support vector machine, regularized neural net, pruned classification tree, or AdaBoost). Dynamic classification and regression are done by imposing a prior to characterize system dynamics. Depending on the prior, the resulting model may be called a hidden Markov model, finite state transducer, dynamic Bayesian network, or conditional random field (dynamic classification), or a Kalman filter, extended Kalman filter, or switching Kalman filter (dynamic regression).

8:45

4aSP2. Joint position/wave number and time/frequency features of signals. Leon Cohen (Dept. of Phys., Hunter College, 695 Park Ave., New York, NY 10021, leon.cohen@hunter.cuny.edu) and Patrick Loughlin (Univ. of Pittsburgh, Pittsburgh, PA 15261)

Joint position/wave number and time/frequency representations of nonstationary signals and noise yield features that may be used to classify signals. Among these are instantaneous frequency, group delay, instantaneous bandwidth, and other conditional, or local, moments. We present a review of these concepts and give examples from various fields. We also discuss how these features change when a pulse propagates through a dispersive medium and show that there are some moment features that are invariant to dispersion. We illustrate via simulation the utility of these dispersion-invariant features for classification of man-made objects (steel shells) in shallow water environments. Finally, we show how to transform equations of motion into phase space, and the advantages of this transformation, in terms of methods of approximation and characterization of signals and nonstationary noise. [Work supported by ONR.]

9:25

4aSP3. Testing for periodicity in speech waveforms. Betul Arda, Daniel Rudoy, and Patrick J. Wolfe (Statist. and Inf. Sci. Lab., Harvard Univ., Oxford St., Cambridge, MA 02138, patrick@seas.harvard.edu)

Speech waveform segments can roughly be categorized as voiced or unvoiced, in accordance with periodicity properties of the glottal source, and inferring this classification from data is in turn an important task underlying a variety of speech classification problems. This presentation describes a formal hypothesis testing framework for the detection of periodicity in general acoustic sources, with application to online voiced/unvoiced segmentation of speech signals. Beginning with the classical approach of Fisher, a variety of test statistics are proposed and analyzed in this context. Asymptotic analyses are provided, along with simulations to demonstrate the efficacy of such methods in the presence of compound periodicities, harmonic structure, and the low signal-to-noise-ratio environments typical of real-world speech applications. [Work supported in part by DARPA, NGA, and NSF.]

9:45

4aSP4. Entropy estimation using pattern matching in bioacoustic signals. John R. Buck (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., N. Dartmouth, MA 02747, johnbuck@ieee.org) and Ryuji Suzuki (MIT, Cambridge, MA 02139)

Many bioacoustic signals consist of a sequence of discrete stereotyped sounds occurring in repeated patterns. A natural question to ask is how best to characterize the underlying structure of the source producing the sequence of sounds. The structure of the source manifests itself as constraints on the patterns observed in the sequence of sounds. These constraints determine how predictable the order of the sounds is. The information entropy of a discrete symbol sequence is a quantitative measure of how unpredictable the sequence is. A straightforward but biased technique for estimating the entropy of an unknown source is to substitute observed symbol frequencies into parametric models such as Markov models. More general nonparametric entropy estimators exploit the relationship between the entropy and the average length of matching patterns within the sequence. This nonparametric entropy estimate forms an upper bound on the amount of information conveyed by the sequence of sounds. Additionally, comparing entropy estimates from the parametric and nonparametric models provides a hypothesis test determining whether the parametric model sufficiently captures the constraints of the source. These techniques are illustrated in analyses of humpback whale songs and leopard seal calling bouts.

10:05—10:15 Break

Contributed Papers

10:15

4aSP5. Classification of audio signals using generalized spatial fuzzy clustering. Huynh V. Luong (Univ. of Ulsan, Korea, huynhlvd@yahoo.com), Cheol Hong Kim (Chonnam Natl. Univ., Korea), and Jong-Myon Kim (Univ. of Ulsan, Korea)

With the increasing use of multimedia data, the need for automatic classification and retrieval of certain kinds of audio data has become an important issue. In this paper, we propose an efficient method of audio signal segmentation and classification from audiovisual database. While conventional methods apply thresholding to audio features such as energy and zero-crossing rate to detect the boundaries, causing misclassification for audio signals which contain certain audio effects such as fade-in, fade-out, and cross-fade, the proposed algorithm, called general spatial fuzzy c-means algorithm (GSFCM), solves the problem by taking into account the local spatial information which is weighted correspondingly to neighbor elements based on their distance attributes. GSFCM detects the boundaries between two different audio signals, classifies segments, and then extracts unique feature vectors. This results in the accurate detection and classification. Experiment results for the audio signal from TV news program at 44.1 kHz with 30-min long confirm that the proposed method outperforms conventional methods in terms of accuracy of the audio signal classification. These results demonstrate that the proposed method is a suitable candidate for audio-video indexing which is compressed by MPEG. [Work supported by the MKE, Korea, under the ITRC supervised by IITA (IITA-2008-(C1090-0801-0039)).]

10:30

4aSP6. Detection and classification of defects in underground pipes using reflection coeffecients. Muhammad Tareq Bin Ali (School of Eng., Univ. of Bradford, BD7 1DP, UK, mtbinali@bradford.ac.uk)

Detection and classification of defects developed in underground pipes have been an important issue for a long time. A novel sensor has been developed to find defects (i.e., blockage, deformation, crack, etc.) in buried pipes. Theoretical analysis has been supported by the experimental results. The sensor consists of a speaker and four microphones. The location of the defect is determined by the reflected signal recorded in the microphones. The reflected signal from a particular defect is sent through filter banks to identify the signal properties, which are related to the property of the defect. Different types of defects have been simulated in the lab and in the field. The sensor has been found to locate and classify the defect considerably well.

10:45

4aSP7. Detecting and localizing pipe changes via matched field processing. A. Tolstoy (ATolstoy Sci., 1538 Hampton Hill Circle, McLean, VA 22101, atolstoy@ieee.org), K. Horoshenkov, M. T. Bin-Ali, and S. J. Tait (Univ. of Bradford, Bradford, UK)

In this work we shall demonstrate that the presence of some pipe changes, e.g., blockages can be detected via matched field processing given an array of two or more microphones. In particular, the location of such transient irregularities can also be determined using this technique. We shall show data processed with this technique for a variety of blockages in a variety of locations for a variety of pipes (concrete, PVC, and clay). There is no acoustic modeling of the fields required, and only the simple linear processor is applied. The method can be used to determine the location of change and its severity.

THURSDAY MORNING, 21 MAY 2009 PAVILION WEST, 7:30 TO 10:50 A.M.

Session 4aUWa

Underwater Acoustics and Structural Acoustics and Vibration: Monostatic and Bistatic Detection of Elastic Objects Near Boundaries: Methodologies and Tradeoffs I

Mario Zampolli, Cochair
NATO Undersea Research Ctr., 19138 La Spezia, Italy

Karim G. Sabra, Cochair
Dept. of Mechanical Engineering, Georgia Inst. of Technology, Atlanta, GA 30332-0405

Chair's Introduction—7:30

Invited Papers

7:35

4aUWa1. Interpreting free-field scattering by metallic truncated cylinders and spherical shells. Philip L. Marston (Phys. and Astron. Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Kyungmin Baik (Inst. of Sound and Vib. Res., Highfield, Southampton S017 1BJ, UK)

When investigating the complications to the scattering caused by proximity of a target to a boundary, a detailed ray-based interpretation of free-field scattering is often helpful. For tilted metallic cylinders having flat ends, when ka exceeds about 10, strong contributions to the backscattering have been identified with the radiation of sound by guided elastic waves associated with meridional rays, helical rays, and face-crossing rays [K. Gipson and P. L. Marston, J. Acoust. Soc. Am. **107**, 112–117 (2000)]. These features also affect the frequency response as a function of tilt and bistatic synthetic aperture sonar (SAS) and acoustic holographic images [K. Baik, Ph.D. thesis, WSU, 2008]. When considering the response of shells to transient insonification, it can also be helpful to construct the time-frequency response and to interpret the response using ray-based methods [D. H. Hughes, Ph.D. thesis, WSU, 1992; S. F. Morse and P. L. Marston, J. Acoust. Soc. Am. **111**, 1289–1294 (2002)]. Correct interpretation of the time-frequency response requires detailed consideration of the dispersion and attenuation of elastic waves guided by the shell and the shape of the radiated wavefronts. When considering low-frequency scattering, the inertia of the target significantly influences the scattering. [Work supported by ONR.]

7:55

4aUWa2. Boundary effects on elastic signatures of proud tilted aluminum cylinders. Jon La Follett and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, lafollej@mail.wsu.edu)

Synthetic aperture sonar (SAS) and frequency response as a function of tilt (acoustic color) are two methods used to study target backscattering. Close proximity to a boundary can affect both the SAS and spectral signatures of a target. To improve understanding of these effects, scaled tank experiments were carried out on solid aluminum cylinders having flat ends and length to diam ratios of 2:1 and 5:1. To partially simulate the mechanisms present when an object is resting on the ocean bottom and illuminated at shallow grazing incidence, the cylinders were suspended through the air-water interface of a tank. Monostatic measurements were obtained as the source /receiver was scanned along a line parallel to the interface to produce SAS images. Backscattering measurements were also made as the target was rotated in a plane parallel to the interface, with the source/receiver stationary, to give the spectrum (color) as a function of tilt. Some of the elastic features in the SAS images and the acoustic color diagrams could be interpreted using a previously developed ray-based theory of generalized Rayleigh waves [K. Gipson and P.L. Marston, J. Acoust. Soc. Am. **106**, 1673–1689 (1999);, **107**, 112–117 (2000)]. [Research supported by ONR.]

8:15

4aUWa3. Monostatic and bistatic measurements of targets resting on or buried under the seafloor. Joseph Lopes (Naval Surface Warfare Ctr.-Panama City Division, Panama City, FL 32407-7001, joseph.l.lopes@navy.mil), Kevin Williams, Steve Kargl, Todd Hefner, Eric Thorsos (Univ. of Washington, Seattle, WA 98105), Philip Marston (Washington State Univ., Pullman, WA 99164), Iris Paustian, and Raymond Lim (Naval Surface Warfare Ctr.-Panama City Division, Panama City, FL 32407-7001)

Measurements were conducted to investigate monostatic and bistatic scattering of targets resting on or buried under a seafloor. The measurements were performed in NSWC-PCD's Facility 383, which is a 13.7 m deep, 110 m long, 80 m wide test pool with a 1.5 m layer of sand on the bottom. Two synthetic aperture sonar (SAS) rail systems were utilized in the measurements, and they were placed perpendicular to each other and oriented so as to look at the same region of the bottom. This strategy allowed a particular target and environment configuration to be set up and studied for both monostatic and bistatic geometries. Targets included a 0.61 m diam stainless steel shell and two 0.3 m diam solid aluminum cylinders with lengths of 0.61 and 1.52 m. This paper summarized the measurement setup and instrumentation. A sample of the results of these measurements will be presented. Subsequent papers will provide detailed analysis and comparisons with predictions of models. [Work supported by the Office of Naval Research and the Strategic Environmental Research and Development Program.]

8:35

4aUWa4. Measurement and modeling of solid cylinders near interfaces. Kevin Williams (College of Ocean and Fishery Sci., Univ. of Washington, 1013 NE 40 St., Seattle, WA 98105, williams@apl.washington.edu), Joe Lopes (Naval Surface Warfare Ctr.–Panama City, Panama City, FL 32407), Eric Thorsos (Univ. of Washington, Seattle, WA 98105), and Philip Marston (Washington State Univ., Pullman, WA 99164)

Acoustic scattering from solid cylinders located near interfaces include effects due to energy interacting with those interfaces. Therefore, modeling cylinder response also requires models of scattering from and penetration across those interfaces. The simplest modeling can be carried out using a plane wave approximation. Using this approximation finite element results for a solid cylinder in the freefield are used to calculate the acoustic scattering of the same cylinder located near an interface. These calculations are compared to experimental data for cylinder target strength and possible reasons for differences seen are discussed. The physical mechanisms responsible for the cylinder's response are examined and cylinder surface displacements are shown. [Work supported by the Office of Naval Research and the Strategic Environmental Research and Development Program.]

8:55

4aUWa5. Modeling the low- to mid-frequency scattering from a proud cylinder: Boundary and near-field effects. Mario Zampolli (NURC, Viale San Bartolomeo 400, 19126 La Spezia, Italy) and Kevin L. Williams (Appl. Phys. Lab., College of Ocean and Fishery Sci., Univ. of Washington, Seattle, WA 98105)

A 2:1 aspect ratio solid aluminum cylinder is placed on the planar interface between two fluids (water/sand and water/air), with a point source radiating at frequencies for which the acoustic wavelengths range from 0.5 to 12 cylinder lengths. The distance between the source and the target is approximately 100 wavelengths, relative to the center frequency, and a vertical receive array is placed near the source. The problem is studied using a finite-element target scattering model, which is capable of treating axially symmetric objects via the decomposition of the unknowns into a Fourier basis around the axis of symmetry. Since the axis of the cylinder is parallel to the planar interface, the overall problem is not axially symmetric. Nevertheless, an approximate solution is obtained, which takes into account the interface reflection of the incident field, as well as the first bounce of the scattered field (via the Helmholtz–Kirchhoff integral with layered medium Greens functions). Small variations in the source and receiver positions cause large changes in the received signal, which can be explained by a Lloyd mirror-like interference resulting from the coherent addition of the point-sources and image point-sources with which the incident field and the scattered field can be described.

9:15

4aUWa6. Acoustic color of elastic objects near boundaries: High-fidelity, high-speed, 3-D finite-element modeling. David S. Burnett (Naval Surface Warfare Ctr., Panama City Div., 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil)

NSWC PCD has developed a computer simulation system for modeling the acoustic color (target strength versus frequency and aspect angle) of realistic 3-D objects that are near to or straddling the interfaces between different fluids. It employs high-fidelity finite-element modeling of acoustic scattering from elastic objects (fully 3-D physics throughout object and environment), implemented in a custom-designed, scalable-architecture, multiblade rack system that efficiently manages the modeling of different parts of the frequency spectrum. The system automatically runs hundreds of thousands of finite-element models, dynamically changing the mesh resolution and outer fluid boundaries of the models as they sweep over frequency, and it produces a variety of outputs, the principal one being an acoustic color contour plot. This paper will begin with a brief overview of the acoustic color simulation system, followed by the results of two experimental validations of the system: (1) scattering from an aluminum cylinder in free space, and (2) scattering from an aluminum cylinder straddling the interface between two different fluids. The simulated acoustic color results will be compared with experimental, numerical, and theoretical results presented in other papers in this session. [Work supported by ONR and SERDP.]

9:35

4aUWa7. Modeling of acoustic scattering by sphere on a planar seabed. Zhengliang Cao, Shuanping Du, Shihong Zhou, and Fangyong Wang (Hangzhou Appl. Acoust. Res. Inst., Hangzhou 310012, China)

A model of acoustic scattering from spherical target above a planar seabed is advanced to a condition of irradiation by a point source, with both of T matrix method and complex images method. Compared to the other model [J. A. Fawcett and R. Lim, J. Acoust. Soc. Am. **114**, 1406–1415 (2003)], this model could be used to calculate scattering field from target above a planar interface in three dimension space or by bounded source beam. Comparing some coefficients by complex images method with that by analytical formula or numerical quadrature, the computing method of the model is examined to be efficient and accurate. In addition, numerical examples of a rigid sphere and an elastic spherical shell are compared for scattering field from the target on a fluid seabed, and the scattering field dependent angle is investigated from the different of grazing angle of incident wave. [Work supported by the National Natural Science Foundation of China (Grant No. 10704068).]

9:50—10:05 Break

10:05

4aUWa8. Three-dimensional structural-acoustics modeling and its validation for free-field and littoral environments. Saikat Dey (Global Strategies Group (North America), 2200 Defense Hwy., Ste. 405, Crofton, MD 21114, saikat.dey.ctr.in@nrl.navy.mil), Angie Sarkissian (Naval Res. Lab., Washington, DC 20375-5320), Eris S. Mestreau (Global Strategies Group (North America), Crofton, MD 21114), Brian H. Houston (Naval Research Lab., Washington, DC 20375-5320), and Larry Kraus (Global Strategies Group (North America), Crofton, MD 21114)

Accurate numerical modeling of structural acoustics scattering in littoral environments, in mid-to-high frequency regimes, presents several challenges, including the ability to model the truncation of the nonhomogeneous exterior domain with free surfaces. Additionally, the frequency-dependent nature of the problem introduces significant dispersion errors with increasing frequency. This makes low-order three-dimensional discretization prohibitively expensive for high accuracy computations. We present a modeling framework for three-dimensional acoustic scattering in littoral environments treated as two half-spaces defined as the fluid and the sediment, respectively. The fluid portion may terminate at a free-surface boundary. The sediment may be a damped fluid or poroviscoelastic solid. The model admits rigid or viscoelastic scatterer with complex three-dimensional shapes including internal structural details and fillings. Exterior domain truncation uses perfectly matched-layer (PML) approximations. An hp-finite-element approximation scheme is utilized to control dispersion errors providing high-accuracy solutions in mid-to-high frequency regimes. Large-scale three-dimensional problems consisting of millions of unknowns solved scalably using a combination of efficient multifrontal solvers and domain-decomposition (FETI-DP) techniques. Numerical results validating applications in free field as well as littoral environment against experimental data will be presented. [Work supported by HPCMP and ONR.]

10:20

4aUWa9. Coherent space-time-frequency processing to enhance the bistatic detection and classification of elastic shells in shallow water. Shaun D. Anderson and Karim G. Sabra (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332-0405)

For underwater sonar, time-frequency analysis has been shown to be a powerful tool for detection and classification of man-made targets. For instance, with traditional monostatic systems, a key energetic feature of spherical shell is the coincidence pattern, or midfrequency enhancement, that results from antisymmetric Lamb-waves propagating around the circumference of the shell. The development by the Navy of mine countermeasure sonar systems, using a network of autonomous systems, provides a mean for multiple bistatic measurements, and thus potentially bistatic enhancement for target detection. However, time-frequency representations of bistatic simulations of scattered signals from spherical shell show that this coincidence pattern typically shifts in both time and frequency with respect to the monostatic case. Hence, this time-frequency shift is challenging for bistatic target detection algorithms based on standard array processing techniques. Design of robust multistatic sonar system based on the generalized space-time-frequency coherence of the bistatic measurements will be discussed. The influence of the source-receiver configuration and interface reflections on the proposed approach have been investigated numerically and experimentally using data collected in shallow water with an elastic spherical shell [Work supported by ONR Code321, N000140810087.]

10:35

4aUWa10. Experimental investigation of bottom target detection by single channel iterative time reversal. Yingzi Ying, Shengming Guo, Bingwen Sun, Li Ma (Inst. of Acoust., Chinese Acad. of Sci., 21 Beisihuanxi Rd., Beijing 100190, China, yingyz05@mails.gucas.ac.cn), Zhenghua Cai, Huamin Fu, and Bing Hu (Haisheng Technol. Ltd., Yichang 443003, China)

Iterative time reversal process will gradually lead echo waves to converge to a dominant narrowband resonant mode of the target and enhance the return level in noisy and reverberant environment. This technique is used in bottom target detection and an experiment has been performed in the Yellow Sea, China. The experiment is in a monostatic configuration, and the target, which is a 53 cm external diameter and 260 cm long stainless steel cylindrical shell with concrete interior, is resting on the seafloor, and the directional transceiver, which is a transmitter and receiver couple, is located right above the target. First, a broadband interrogation pulse is launched, and the echo is measured and a bandpass filter is applied to avoid transceiver response peak, then the signal is time reversed and retransmitted, and repeat above procedures iteratively. The bottom reverberation will gradually be suppressed, and the center frequency of converged signal corresponds to a target resonance frequency, which is different from inhomogeneous bottom response in no target case. The existence of target is determined by this important acoustic signature, and the results illustrate the feasibility of this method. [Work partially supported by the CAS Innovation Fund.]

Session 4aUWb

Underwater Acoustics and Signal Processing in Acoustics: Waveguide Invarient Principles for Active and Passive Sonars

Altan Turgut, Cochair
Acoustics Div. Code 7120, Naval Research Laboratory, Washington, DC 20375

Lisa M. Zurk, Cochair
Electrical and Computer Engineering Dept., Portland State Univ., Portland, OR 97207

Chair's Introduction—8:00

Invited Papers

8:05

4aUWb1. An overview of the waveguide invariant and some of its applications to signal processing. W. A. Kuperman, G. L. D'Spain, Peter Gerstoft, W. S. Hodgkiss, H. C. Song, and A. M. Thode (Marine Physical Lab., Scripps Inst. of Oceanogr., U.C. San Diego, La Jolla, CA 92093-0238)

While the waveguide invariant is a compact descriptor of certain aspects of acoustic propagation in the ocean, it also has a broad range of applications from describing acoustic fluctuations to assisting in various types of signal processing. After presenting a brief theoretical and experimental overview of the invariant in terms of both normal modes and rays, some applications to signal and array processing are discussed.

8:35

4aUWb2. Effect of shallow water variability on the waveguide invariant distribution. Daniel Rouseff (Appl. Phys. Lab., College of Ocean and Fishery Sci. Univ. of Washington, Seattle, WA 98105) and Valery Petnikov (Russian Acad. of Sci., Moscow 119991, Russia)

When mapped versus range and frequency, acoustic intensity often displays a regular pattern of striations, ribbons of high intensity. The trajectory of these striations may be described by the waveguide invariant, commonly designated as beta. While beta may be formally an invariant quantity, it is not necessarily a constant, particularly in highly variable shallow water environments. To study fluctuations in beta due to fluctuations in the environment, it is useful to generalize from an invariant scalar to the waveguide invariant distribution. Through data analysis and simulation, the waveguide invariant distribution is calculated for different locations including the New Jersey shelf off the United States and the Kamchatka shelf off Russia. The effects of isotropic and anisotropic internal waves are quantified. [Work supported by ONR and the Russian Foundation for Basic Research.]

8:55

4aUWb3. Waveguide invariant motion compensation for adaptive beamforming. T.C. Yang and Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375)

Performance of high-resolution adaptive beamformers is significantly degraded when moving sources and interferers are considered. For a source changing bearing, the motion effect can be compensated by integrating over beam covariant matrix. For a source changing range, the motion effect can be compensated by a frequency change using waveguide invariant theory [T. C. Yang, J. Acoust. Soc. Am. **113**(1), 245–260 (2003)]. Simulated data showed that the signal beam of a moving source is narrow as for a stationary case when motion compensation is applied. Also, interference power has been suppressed by 6–10 dB when the motion of the interferer does not match that of the source. Motion compensation algorithms were also applied to ship noise data from the passing merchant ships, collected on a bottomed horizontal array during the RAGS 2003 experiment at the New Jersey Shelf. Similar results are obtained indicating narrower beamwidths and selective suppression of signals from the passing merchant ships when proper motion compensation is considered. [Work supported by ONR.]

9:15

4aUWb4. Passive striation based geo-acoustic inversion using ships of opportunity. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

The coherent interaction of acoustic multipaths leads to an interference pattern in range. As range changes, the locations of the peaks (and troughs) of this interference pattern shift (as the group velocities of the modes change) in a sometimes predictable fashion. The intensity pattern of a moving source, plotted as a function of range and frequency, is commonly referred to as striation patterns. These patterns are commonly observed in the passing of a loud surface vessel in the spectrogram of a single hydrophone. In this paper, a geoacoustic method is presented based upon the quantitative measure of striation slope (wave guide invariant parameter) and the striation spacing in frequency (reciprocal time spread). Both of these parameters are generally sensitive to the geo-acoustic parameters of the sediment. In 2003 this technique was applied to a passing surface ship [K. Heaney, IEEE JOE **29**(1), 43–50 (2003)] and to a range-

dependent benchmark case [K. Heaney, IEEE JOE **29**(1), 88–99 (2003)]. The approach in these papers compared the measured acoustic observables with those from a table predicted by the forward computation using the broadband PE. Current implementation of the algorithm uses a Gaussian beam code to efficiently compute the acoustic observables.

9:35

4aUWb5. Waveguide invariance for active sonar. Jorge E. Quijano and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201)

Active sonar signal processing in shallow water has proven to be a challenging problem due to the strong interaction of sound with the boundaries of the channel and the dependence on typically unknown environmental parameters. This has motivated research on properties of acoustic propagation that are not sensitive to those factors, such as the waveguide invariance. The invariance principle has found application in passive sonar signal processing by relating the frequency content of a broadband source to the range between source and receiver. More recently, experimental evidence has suggested that a similar structure exists for active sonar. This structure provides additional information about the location of a target, and information can be exploited in sonar processing algorithms such as target tracking. Data from several sea experiments have been analyzed to determine the behavior of the active invariance, and tank experiments have been designed to confirm the presence of the range-frequency structure in signals reflected by a moving target. This presentation provides an overview of the active invariance phenomena and describes the performance of a target tracking formulation that incorporates invariance structure into the state space representation for improved performance.

9:55

4aUWb6. Environmentally invariant features for classification of active sonar signals. Patrick Loughlin (Depts. of Bioengineering and ECE, Univ. of Pittsburgh, 745 Benedum Hall, Pittsburgh, PA 15261, loughlin@pitt.edu) and Greg Okopal (Univ. of Pittsburgh, Pittsburgh, PA 15261)

Dispersion and damping (frequency-dependent spreading and attenuation) can be significant in shallow water sound propagation. These propagation-induced effects can be detrimental to classification of active sonar returns because the observed backscatter depends not only on the target, but also on the propagation environment and how far the wave has traveled, resulting in increased variability in the received signals. We address this problem by extracting propagation-invariant features from the wave that can be used in an automatic classifier. In this talk, we review various moment-like features we have developed that are invariant per mode to dispersion and damping. Simulations of the backscatter from different steel shells propagating in a channel with dispersion and damping are presented to demonstrate the classification utility of the various invariant features. [Work supported by ONR Grant N00014-06-1-0009.]

10:15—10:30 Break

10:30

4aUWb7. Wavguide invariant-based characterization of wideband active sonar clutter discretes. Ryan Goldhahn, Jeffrey Krolik (Duke Univ., ECE Dept. Hudson Hall 130, P.O. Box 90291, Durham, NC 27708, rag15@ee.duke.edu), and Charles W. Holland (Penn State Univ. Appl. Res. Lab, Appl. Sci. Bldg., State College, PA 16804)

In active sonar, clutter discretes can produce strong, target-like returns which often produce false alarms of water column targets. While false alarm reduction methods based on statistical feature-based classifiers often lack sufficient training data, matched-field based classifiers often suffer from model mismatch. A waveguide invariant-based approach for estimating the magnitude short-time Fourier transform (STFT) of reverberation returns was presented [Goldhahn *et al.*, J. Acoust. Soc. Am. **124**(5), 2841–2851, (2008)]. In this paper, the waveguide invariant properties of the reverberation are used to predict the frequency selective fading of strong clutter discretes. In particular, comparison of waveguide invariant-based magnitude STFT estimates are compared with predictions made using a model of frequency-selective fading from a clutter discrete. The results are further compared with real sonar returns collected during the SCARAB98 experiment off a shipwreck in the Malta Plateau. The results suggest that when accurate characterization of the geoacoustic environment are available, the estimated STFT magnitude spectrum obtained by using a propagation and scattering model compare favorably with those obtained by averaging the reverberation along waveguide invariant striations. [Work supported by ONR.]

Contributed Papers

10:50

4aUWb8. Investigation of bistatic invariance principles for active sonars. Altan Turgut and Roger Gauss (Naval Res. Lab., Acoust. Div., Washington, DC 20375)

Application of bistatic invariance principles to mid-frequency active sonar systems is investigated using data collected during two recent experiments conducted at the Malta Plateau and East China Sea. Low-frequency (350–650 Hz) and mid-frequency (1.5–3.5 kHz) LFM pulses were transmitted and direct-blast and return signals from an echo-repeater and several strong scatterers simultaneously recorded on a towed array, a drifting volumetric array, and a moored vertical line array. At low frequencies, application of bistatic invariance principles to target detection is demonstrated. At mid-frequencies, the spectrograms of the direct-blast signals showed regular striation patterns that were used to estimate the waveguide invariant parameter beta. However, both measured and simulated spectrograms of the signals scattered from an oil rig indicated the complexity of the striation patterns. More complex patterns of striations in the measured spectrograms might be due to azimuthal dependency of the scattering kernel, source /receiver motion, and low SNR. [Work supported by the Office of Naval Research.]

11:05

4aUWb9. Full spectrum acoustic wave propagation prediction. Cathy Ann Clark (Naval Undersea Warfare Ctr., 1176 Howell St., B1320, Rm 457, Newport, RI 02841, cathy.clark@navy.mil) and Kevin B. Smith (Naval Postgrad. School, Monterey, CA 93943)

An overview of a normal mode solution to the Helmholtz wave equation to describe the underwater sound field for a fixed point source in a plane multilayered medium which utilizes Bessel functions of order 1/3 is presented. The mode functions are continuous across turning points of the separated depth-dependent differential equation due to careful selection of

the representations to be used for Bessel function arguments in various regions of the complex plane. A quotient involving vertical wave number and phase is seen to behave as a constant through turning points, enabling the mode amplitude functions to remain analytic, changing from oscillatory to exponential on traversing the turning point, thus enabling smooth incorporation of the continuous spectrum. The method also provides vertical directionality at all field points without post-processing the complex acoustic field. Comparisons of model results to a limited number of measured data sets and benchmark propagation codes are presented. Derivation and verification of the solution for bottom-interacting modes, including shear and compressional reflection and transmission for a layered bottom, as well as an extension into horizontally varying, shallow water environments are also discussed. Portions of this work have been published in the IEEE Journal of Oceanic Engineering.

11:20

4aUWb10. Acoustic ranging and waveguide invariant parameter estimation using virtual arrays. Altan Turgut (Naval Res. Lab., Acoust. Div., Washington, DC 20375)

A method for estimating the range of an unknown broadband acoustic source in a waveguide [Thode *et al.*, J. Acoust. Soc. Am. **108**(4), 1582–1594 (2000)] is revisited and extended to estimate both waveguide invariant parameter "beta" and source range in shallow water. In the new method, two or more vertical arrays are used without requiring a signal sample from a guide-source. It was shown that both methods are mathematically identical and they both provide robust range estimation even when the reference signal sample is used from different time and/or different vertical array location. It was also demonstrated that an image processing tool, Hough Transform method, provides robust parameter estimation from virtual array output data. In addition, the parameter estimation method was validated under both summer and winter conditions by using incoherent noise data to localize and track merchant vessels and to estimate waveguide invariant parameter. [Work supported by the Office of Naval Research.]

11:35

4aUWb11. Passive ranging using the waveguide invariant. Kevin L. Cockrell and Henrik Schmidt (Dept. of Mech. Eng., Massachusetts Inst. of Technol, Rm. 5-204, 77 Massachusetts Ave., Cambridge, MA 02140, cockrell@mit.edu)

A range versus frequency spectrogram of an acoustic field due to a fixed source in a waveguide will exhibit striations whose slopes depend on the range to the acoustic source and the value of the waveguide invariant. While many authors have pointed out that the range to an acoustic source can be estimated from the slopes of the striations in the spectrogram, few have presented an explicit algorithm to do so. An algorithm for estimating the range is presented and tested on experimental data collected in a shallow water waveguide during GLINT08, an exercise performed off of Pianosa Island, Italy. The experimental data consist of a fixed broadband acoustic source emitting energy at frequencies from 300 to 800 Hz, with a hydrophone measuring the acoustic field along a 1.75-km track directly away from the acoustic source.

THURSDAY AFTERNOON, 21 MAY 2009 GRAND BALLROOM II, 1:00 TO 3:30 P.M.

Session 4pAAa

Architectural Acoustics: Acoustics of Health and Healing Environments

Kenneth P. Roy, Chair
Innovation Ctr., Armstrong World Industries, 2500 Columbia Ave., Lancaster, PA 17604

Chair's Introduction—1:00

Invited Papers

1:05

4pAAa1. Acoustical designs in a new children's hospital. Francis Babineau, Jr. (Johns Manville, 10100 W. Ute Ave., Littleton, CO 80127, francis.babineau@jm.com)

The importance of noise control and acoustic comfort in healthcare facilities has been well documented. This issue is even more critical in pediatric facilities, given the often frightening and stressful circumstances associated with a child being in a hospital. Recently, The Children's Hospital in Denver, CO constructed a new facility in neighboring Aurora, CO. The new facility was opened in Oct. 2007, and one of the design goals was to improve the acoustic environment by implementing evidence-based design strategies. However, one of the challenges in improving hospital acoustics is to do so without introducing additional infection control risks. As part of the project, a series of noise measurements were performed at the old hospital and in analogous locations in the new hospital, after the new hospital was occupied. This paper will present the results of the noise measurements and discuss the impact (positive and negative) of various design elements on the acoustic environment.

1:25

4pAAa2. Perceptions and expectations of speech privacy in healthcare environments. Kenneth Good (Acoust. Privacy Enterprises, LLC, P.O. Box 252, Mount Joy, PA 17552) and Nikki Rineer (Hope Within, 4748 Harrisburg Pk., Elizabethtown, PA 17022)

Most methods for evaluating speech privacy were developed for offices and corporate environments and from the point of view of productivity and distraction impacts on the listeners. How do these methods translate to healthcare and other environments where confidential containment of information is required by law? This case study will look at the objective measurements of speech privacy along with patient subjective impressions and expectations surveyed.

1:45

4pAAa3. Evaluation and control of the acoustical environment in a long-term-care facility. Murray Hodgson and Gavin Steininger (Acoust. and Noise Res. Group, SOEH-MECH, Univ. of British Columbia, 3rd Fl., 2206 East Mall, Vancouver, BC, V6T1Z3 Canada)

This paper discusses the acoustical evaluation of the Minoru Residence long-term-care facility, to respond to concerns by its staff regarding the acoustical conditions. A review of existing acoustical standards with an analysis of their applicability to health-care facilities was conducted for the problems observed in the Minoru Residence. Measurements were made of the acoustical characteristics: unoccupied and occupied noise levels, reverberation times, Speech Intelligibility Index, and noise isolation. They showed that background noise levels in several key areas including the Rehabilitation Office and the Patient Lounges exceeded acceptable values. Reverberation times were excessive in the entrance lobby and patient common areas. The Speech Intelligibility values in the Nurse Stations and Rehabilitation Offices were below acceptable values. The noise isolation was inadequate between the entrance lobby and office areas. Recommendations were made for the improvement of the acoustical conditions. These recommendations include the reinforcement of the Front Office façade, and the application of acoustical ceiling tiles to the Rehabilitation Office and the entrance lobby.

2:05

4pAAa4. Achieving green design acoustical standards in healthcare facilities. Peter Holst and Ethan Salter (Charles M. Salter Assoc., 130 Sutter St., Ste. 500, San Francisco, CA 94104, peter.holst@cmsalter.com)

With the advent of the green guide for healthcare, and the introduction of LEED health care, which both include acoustical design credits, the benefits of good acoustical design have been recognized for improving patient recovery rates as well as staff health and efficiency. The acoustical issues in designing health care facilities cross the spectrum of acoustical design: sound isolation/speech privacy, room acoustics, sound masking, mechanical noise/vibration, environmental noise, and project-specific features, such as MRI and helipads. It is the responsibility of the acoustical consultant to address each item with the design team to develop cost-effective solutions to meet the criteria established by the green standards. This paper will provide background information on the acoustical design challenges for healthcare facilities and will discuss general approaches that achieve the goals of green acoustical design. Specific examples from projects will also be included, drawing from experience in design of hundreds of health-care facilities, as well as documentation of credits achieved for numerous medical centers that plan to submit for LEED healthcare. This paper will be a valuable resource for acoustical consultants in the design of all types of healthcare facilities, such as hospitals, central plants, medical office buildings, and nursing facilities.

2:25

4pAAa5. Safe and sound? Sleep disruption in healthcare facilities. Joanne Solet (Psychiatry, Cambridge Health Alliance, Cambridge, MA 02138, joanne_solet@hms.harvard.edu), Orfeu Buxton (Brigham & Women's Hospital, Boston, MA 02115), Andy Carballeira (Cavanaugh-Tocci Assoc, Sudbury, MA 01776), and Jeffrey Ellenbogen (Massachusetts General Hospital, Boston, MA)

National healthcare quality surveys have found that noise in hospitals is an urgent concern, showing negative impact on patient satisfaction. The purpose of this study was to develop sleep arousal probability threshold curves to specific hospital-based sounds, demonstrating their impact on all stages of human sleep. Recordings were captured of hospital sound sources corresponding to specific categories identified as salient in the American Institute of Architects' Draft Interim Guideline on Sound and Vibration in Healthcare Facilities. Fourteen sounds were calibrated for dynamic presentation through a speaker array at the sleep lab to deliver rising 5 decibel-step exposures from 40 to 70dB(A). Noise-related EEG arousals were quantified using current AASM criteria and summed for each sleep stage by sound type and decibel level to calculate arousal probability threshold curves. The tested stimuli evoked a range of arousal thresholds. At the 50% arousal probability level, stimuli spanned 15 dB(A) Leq in Stage 2 sleep, 17 dB(A) Leq in REM sleep, and 30 dB(A)Leq in Stage 3, the deepest sleep. The findings provide evidence that repeated arousals in all sleep stages occur even in healthy young adults when hospital sounds exceed 45dB(A); responses vary widely by stimulus types.

Contributed Papers

2:45

4pAAa6. Canadian hospital acoustical evaluation. Hind Sbihi and Murray Hodgson (Acoust. Res. Group, SOEH/Mech., UBC, 2206 East Mall, Vancouver, BC V6T1Z3 Canada, murray.hodgson@ubc.ca)

The aim of this study was to evaluate the acoustic conditions of two wards in a research/teaching hospital in British Columbia, Canada. The selection of the wards was based on managerial staff needs and perceptions of issues related to the acoustical working environment with respect to privacy and also aggressive behaviors. The two selected units were an adult emergency department and a long-term care facility where the patients population was a mix of elderly with different mental and physical health conditions. The methods will include long-term noise measurements, building acoustical measurements and interviews with nursing staff. In particular, measurements will be made of the following acoustical parameters in the facilities: unoccupied and occupied noise levels; reverberation time; Speech Intelligibility Index; noise isolation. Results will be evaluated by comparing them with acceptability criteria. Identification of nonoptimal aspects of the facilities acoustical environments will result from the consideration and analysis of staff responses and comparison with published guidelines.

3:00

4pAAa7. A comparison of sound transmission loss on metal stud partitions as the stud configuration changes. Aaron Betit (Veneklasen Assoc., 1711 16th St. Santa Monica, CA 90404)

The draft "Interim sound and vibration design guidelines for hospital and healthcare facilities" drafted by the Joint Subcommittee on Speech Privacy of the ASA recommends STC 50 partitions between exam rooms with no masking sound provided. These partitions are typically constructed of multiple layers of gypsum board on single steel studs, which are widely believed to achieve the required ratings based on published test reports. However, virtually all laboratory testing is with 25 gauge studs 24 in. on center, whereas with a 15 ft floor to floor height typical of hospitals, 16 in. gauge studs installed 16 in. on center are often required for structural reasons. There is little published data on the changes in sound transmission loss changes with stud gauge and spacing. A testing program was established, and transmission loss (STC) tests were performed on drywall partitions with various configurations of stud gauges, spacing, and layers of drywall. Measureable decrease in transmission loss as the studs become heavier and as the spacing between studs decreases was measured. The results of the testing program are presented.

3:15

4pAAa8. Noise inside a government owned hospital. Sergio Beristain (Acoust. Lab., E.S.I.M.E., IPN, IMA, P.O. Box 12-1022, Narvarte, Mexico, D. F. 03001, sberista@hotail.com)

A hospital is under evaluation in order to find out the most important noise sources. Noise measurements were carried out within and outside of the hospital, It was found that noise from the neighborhood was not an issue, but noises from within the hospital were loud enough to cause at least some disturbance to the patients and workers. Obviously the larger noise was the one coming from the machinery room, where the laundry, emergency power plant, etc., are located, but also in some of the most important care and treatment rooms, the noise generated inside was well over of the recommended limits for an installation of this type. Some measurement results are presented together with the description of the environment.

THURSDAY AFTERNOON, 21 MAY 2009 — GALLERIA SOUTH, 2:00 TO 4:55 P.M.

Session 4pAAb

Architectural Acoustics, ASA Committee on Standards, and Noise: Indoor Noise Criteria

Lily M Wang, Chair
Architectural Engineering, Univ. of Nebraska--Lincoln, Omaha, NE 68182-0681

Chair's Introduction—2:00

Invited Papers

2:05

4pAAb1. Proposed components of an "ideal" indoor noise criteria rating system. Lily M. Wang (Architectural Engr. Prog., Univ. of Nebraska-Lincoln, Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681)

A number of studies have been conducted at the University of Nebraska on correlating human performance and perception to indoor noise criteria systems. Task performance and subjective perception data were gathered from subjects exposed to background noise conditions commonly due to mechanical systems, including some with tonal components and some with time-varying fluctuations. Results show that perceptions of annoyance and distraction are highly correlated to the character of the noise, but the current indoor noise criteria systems (such as noise criteria and room criteria) do not accurately reflect that relationship. This paper presents a proposal for what an indoor noise criteria rating system should ideally include, to quantify acceptable building mechanical system noise in commercial buildings. [Work supported by the American Society of Heating, Refrigeration and Air-Conditioning Engineers.]

2:30

4pAAb2. A case history comparing noise criteria. Jerry G. Lilly (5266 NW Village Park Dr., Issaquah, WA 98027)

A case history of an indoor HVAC noise problem in a new residential building will be presented. Noise measurements collected in the living room and in the bedroom of the impacted living unit will be examined using several of the available noise criteria methods including, NC, RC, NCB, and RNC (ANSI S12.2). Although the architect and the building owner believed that the HVAC noise was unacceptable, the measured noise levels met the NC, NCB, and RNC noise criteria. Only the RC method was able to accurately detect the problem.

2:55

4pAAb3. Evaluation of mechanical background noise outside the norm of generally accepted criteria. Andrew J. Boone and Michael R. Yantis (Sparling, 720 Olive Way, Ste. 1400, Seattle, WA 98101, aboone@sparling.com)

Cases are presented where mechanical background noise was found to be acceptable by clients, although it was above the threshold of conventional indoor noise criterion. Other instances are shown where noise levels fell within normal limits but were judged unacceptable. Examples include HVAC noise in offices and residences, chiller and other rooftop equipment noise in multifamily dwellings, and pump noise. Sound quality expectations and the perception of noise sources were found to play an important role in the evaluation of these noise sources.

3:20—3:35 Break

3:35

4pAAb4. Using indoor room criteria when the "room" is outside. Byron W. Harrison (TALASKE, 1033 South Blvd., Oak Park, IL 60302, byron@talaske.com)

Indoor noise criteria have applicability in the analysis of outdoor performance spaces. The presentation will provide an overview of the environmental noise issues at the Jay Pritzker Pavilion in Chicago, IL. The project design was largely influenced by noise concerns, in its physical form, audio design strategy, and building systems design. During the postconstruction tuning of the audio system a

number of parameters were adjusted to contend with the unusually low signal-to-noise ratio, including the overall loudness of amplified music, audio signal compression, the content and level of the acoustic enhancement system, and the approach to the active in-house mix of the audio system. An investigation was also undertaken to investigate the continuous and impulsive noise levels and frequency content at various times of day as compared with traditional environmental noise criteria and indoor room criteria. Subsequent studies regarding the impact of nearby construction noise on rehearsals and performances were influenced by the use of indoor noise criteria methods.

4:00

4pAAb5. The room noise criteria (RNC) metric. Paul Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssoc..com)

The recent ANSI S12.2:2008 room noise criteria contains both a survey and an engineering method to specify room noise criteria. The methods use A-weighting and extended NC, respectively. A new metric, titled like the standard, room noise criteria (RNC) is included as a diagnostic tool. It is based on human hearing and more correctly assesses low-frequency sound. In particular, it is sensitive to the standard deviation to random noise and/or low-frequency surging in the 16–125 Hz octave bands such as the sound that can be produced by HVAC systems or other devices. It provides a bridge between the NC and RC criteria by correctly predicting the need for the less stringent (at low frequencies) NC criteria when the HVAC system is well designed (no surging, moderate standard deviation) and also correctly predicting the more stringent (at low frequencies) RC criteria when the HVAC system noise has a large standard deviation and/or surging.

Contributed Papers

4:25

4pAAb6. Correlation of subjective and objective measures of spectral quality. Dakota M. Kelley and Lily M. Wang (Architectural Engr. Prog., Univ. of Nebraska—Lincoln, Peter Kiewit Inst., 1110 S. 67th St., Omaha, NE 68182-0681, dkelley@mail.unomaha.edu)

Common indoor noise criteria with spectral quality indicators have been compared to task performance and subjective noise perception. The analysis seeks to identify a relationship between criteria spectral quality ratings and human perception of common heating, ventilating, and air conditioning background noises. The three criteria evaluated include Balanced Noise Criteria, Room Criteria, and Room Criteria Mark II, due to their inclusion of rumble, hiss, or roar classifications. Study participants worked on typing, math, and verbal tasks while being exposed to various background noise signals, and then completed a questionnaire to describe their perception of the room acoustics. During the data analysis, background noise signals with a non-neutral spectral quality rating were weighted according to their respective spectral indicator, with greater weighting given to lower-frequency signals. Results demonstrate relationships between spectral quality designation and subjective perception of noise fluctuation and tonality. However, a relationship was not found between spectral quality designation and subjective perception of the same parameters (rumble, hiss, or roar). These findings warrant further investigation of the correlation between common criteria ratings and subjective perception. [Work supported by the Univ. of Nebraska Undergraduate Creative Activities and Research Experience Grant.]

4:40

4pAAb7. Acoustical criteria in a two-parameter system for evaluating impact noise insulation. John LoVerde and Wayland Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Experience indicates that impact noise complaints in multi-family joist-framed buildings fall into two broad classes: low frequency thudding from footfalls and mid- to high frequency noise from heel clicks, dragging furniture, etc. The authors have developed a two-parameter system for evaluating impact noise [LoVerde and Dong, J. Acoust. Soc. Am. **119**, 3220 (2006); **120**, 3206 (2006); **122**, 2954 (2007)] that offers considerable improvement over existing metrics (such as FIIC) in terms of both correlation with subjective reaction and comparison of materials intended for improving impact insulation. Based on this system, suggested criteria for impact noise levels are presented. The effects of various design parameters on noise levels are discussed.

Session 4pAB

Animal Bioacoustics: General Topics in Animal Bioacoustics II

David K. Mellinger, Chair
Oregon State Univ., Newport, OR 97365

Contributed Papers

3:00

4pAB1. Equine vocalizations: The start of a search for happiness. David A. Browning (Phys. Dept., Univ. of Rhode Island, 2 Lippitt Rd., Kingston, RI 02881, decibeldb@aol.com), Peter M. Scheifele (Univ. of Cincinnati, Cincinnati, OH 45267-0379), and Rebecca L. Pond (Univ. of Connecticut, Storrs, CT 06269)

As with all perissodactyls, the vocalizations of equines, specifically a horse's whinny, has a variable frequency (or melodic) component as well as just simple tonals. This appears to provide a primitive means of expression, simpler than any song or language but potentially more informative than the purely tonal moos or baahs of cattle or sheep (but a long way from the complexity of some birdsong). Sonograms are compared from in-barn whinnies recorded under apparently stressful (departure of a foal) and pleasant (arrival of the morning feed wagon) circumstances, with the same horse and various horses, to determine if distinctive patterns can be identified. We also compare these with greeting whinnies and departure whinnies. The ultimate goal is to be able to acoustically identify an expression of happiness.

3:15

4pAB2. A vocal repertoire of Asian elephants (*Elephas maximus*) and comparison of call classification methods. Sharon S. Glaeser (Dept. of Biology, Portland State Univ., P.O. Box 751, Portland, OR 97207, sharon@roguetechinc.com), Holger Klinck, David K. Mellinger (Oregon State Univ. and NOAA, Newport, OR 97365), Yao Ren (Marquette Univ., Milwaukee, WI 53201), Patrick J. Clemins (Arlington, VA 22203), Michael T. Johnson (Marquette Univ., Milwaukee, WI 53201), Mandy L. H. Cook, and Randy Zelick (Portland State Univ., Portland, OR 97207)

This study compares classification methods applied to an acoustic repertoire of the Asian elephant (*Elephas maximus*). Recordings were made of captive elephants at the Oregon Zoo in Portland, OR and of domesticated elephants in Thailand. Acoustic and behavioral data were collected in a variety of social contexts and environmental noise conditions. Calls were classified using three methods. First, calls were classified manually using perceptual aural cues plus visual inspection of spectrograms for differentiation of fundamental frequency contour, tonality, and duration. Second, a set of 29 acoustic features was measured for nonoverlapping calls using the MATLAB-based program Osprey, then principal component analysis was applied to reduce the feature set. A neural network was used for classification. Finally, hidden Markov models, commonly used for pattern recognition, were utilized to recognize call types using perceptually-weighted cepstral features as input. All manual and automated classification methods agreed on structural distinction of six basic call types (trumpets, squeaks, squeals, roars, rumbles, and barks), with two call types (squeaks and squeals) being highly variable. Given the consistency of results among the classification methods across geographically and socially disparate subject groups, we believe automated call detection could successfully be applied to acoustic monitoring of Asian elephants.

3:30

4pAB3. The hyena's laugh as a multi-informative signal. Nicolas Mathevon (ENES Lab, Univ. Jean Monnet, Saint-Etienne, France, mathevon@univ-st-etienne.fr) Aaron Koralek, Steve Glickman, and Frederic Theunissen (Berkeley, CA)

Many social mammals use vocalizations to encode information about sex, kinship, individual identity, and morphological cues, as well as motivational and physiological states. In spite of the importance of this multi-informative signaling for the maintenance of social groups, most investigations on information coding in vocal signals have focused on only one cue; e.g., individual identity. Using the opportunity of the captive colony of spotted hyenas *Crocuta crocuta* at the Field Station for the Study of Behavior, Ecology, and Reproduction (University of California, Berkeley), we recorded and analysed the hyena's giggle, one of the most well known calls of this large social African mammal. The acoustic analysis in both temporal and frequency domains was automated using a MATLAB customized routine. The fundamental frequency was tracked using two methods (cepstrum and autocorrelation) followed by a best guess using a Bayesian approach. The differences between giggles from different individuals or groups of individuals were assessed running a multiple analysis of variance (MANOVA in MATLAB), cross-validated by a permuted discriminant function analysis (pDFA, R software). The results show that the hyena's laugh encodes information about age, dominance status, and individual identity, giving to receivers some cues to assess the social position of an emitting individual.

3:45

4pAB4. An intelligent automated apparatus to assess absolute auditory thresholds in the laboratory mouse. Anna Pleuger (Charles Darwin Univ., Darwin, NT, Australia) and Al Yonovitz (The Univ. of Montana, Missoula, MT 59812)

The effectiveness of an intelligent behavioral training and testing apparatus was assessed by using this system to operantly condition mice. The method utilized an infrared grid that determined the location of the mouse and presented reinforcements for auditorily contingent bar-press behavior. The apparatus was fully automated. Absolute auditory thresholds were determined with the descending and ascending method of limits in the same group of C57BL/6 mice. There was a statistically significant difference between thresholds produced by these two methods, with the descending method producing more sensitive auditory thresholds. Thresholds were on average 4.4 dB lower and had smaller standard errors. Overall, the automated apparatus was a highly efficient and precise method for the operant conditioning of mice.

4:00

4pAB5. Social context influences acoustic communication in zebra finches. Clementine Vignal, Julie Elie (Univ. Jean Monnet, Saint-Etienne, France), Hedi Soula (INSERM U870, INSA, Lyon, France), and Nicolas Mathevon (Univ. Jean Monnet, Saint-Etienne, France)

During communication, a signal conveys information between an emitter and a receiver, but indirect receivers can eavesdrop on the interaction. In birds, communication has been demonstrated to often be under the influence of this eavesdropping. Social species show complex communication networks where audience drives individual behaviors. Zebra finches

(*Taeniopygia guttata*) are gregarious songbirds that live in social groups and form life-long pair-bonds. Previous studies showed that the vocal behavior of males highly depends on this audience effect. Males show mate calls preference over other female calls in the presence of an established male-female pair, but not in the presence of unmated male-female or male-male dyads. Males in social groups also show stronger vocal response to female calls than males in social isolation. In this study, we investigate whether female calls of varied social salience evoke differently male calling according to the audience. We show that social context modifies not only call rate in response to female calls of varied social salience, but also acoustic structure of evoked calls. Thus male distance calls are not stereotyped calls whose acoustic cues only convey bird's identity. We propose that fine spectral modifications of the calls could carry information about the emitter motivation.

4:15

4pAB6. Acoustic analyses of two undocumented sound patterns in the *Drosophila suzukii* and *D. takahashii* species subgroups. Yuwen Lai (Dept. of Linguist., Univ. of British Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada), Shu-Dan Yeh (Stony Brook Univ., Stony Brook, NY 11794-5245), Jennifer Gleason (Univ. of Kansas, Lawrence, KS 66045), and John True (Stony Brook Univ., Stony Brook, NY 11794-5245)

Acoustic analyses of the courtship songs of the *suzukii* and *takahashii* subgroups in the *Drosophila melanogaster* species group were conducted. The primary and secondary pulse phases that are common in the *melanogaster* subgroup were not observed in these two subgroups. However, two undocumented sound patterns were discovered. *D. biarmipes* and *D. pulchrella* (*suzukii* subgroup) produce high amplitude, nonrhythmic "toot" sounds, which range from 82–158 ms in duration. The toot sound in *D. biarmipes* has a consistent dynamic frequency profile. It starts with an onset of 479 Hz and gradually falls to 422 Hz and then rises to 477 Hz. The toot sound in *D. pulchrella* has a significantly lower frequency when compared to *D. biarmipes*. Its frequency profile falls gradually from 352–259 Hz. In addition, a "turbo" sound was recorded in *D. prostipennis* (*takahashii* subgroup). It is composed of short, high frequency pulses (520 Hz) with 4 ms interpulse intervals. In the *melanogaster* subgroup, the parameters of pulses have been proposed to play an important role in female preference. The results of the present study suggest that there might be other parameters at play in the species investigated in the current study.

4:30

4pAB7. Recent insights on the mechanisms of frequency discrimination in cicadas (Hemiptera, Cicadoidea). Paulo J. Fonseca (Dept. de Biologia Animal and Centro de Biologia Ambiental, Faculdade de Ciências, Univ. de Lisboa, Bloco C2, Campo Grande, 1749-016 Lisboa, Portugal, pjfonseca@fc.ul.pt) and Axel Michelsen (Univ. of Southern Denmark, DK 5230 Odense M, Denmark)

Mate finding in cicadas is usually mediated by acoustic communication. Males produce a loud acoustic signal that is used to guide females toward singing males. The calling songs are frequently complex with changes in rhythm, amplitude modulation and, in many species, frequency modulation. Therefore, it is likely that the auditory organ encode some of those characteristics and that the nervous system may process the information and extract some species-specific parameters. The tympanic vibrations are transferred to the onion-shaped auditory organ, localized at some distance from the tympanum within the auditory capsule, through a stiff schlerotized apodeme. This configuration has raised problems to the understanding of how the different frequencies of the song, that Fonseca *et al.* (2000) have shown to be finely encoded at the level of the auditory interneurons in the cicada *Tettigetta josei*, are passed on to the auditory organ by the structures of the receptor. Using biophysical, electrophysiological, and anatomical measurements from the receptor structures of the cicada *Tettigetta josei*, a functional model that may allow for the above-mentioned frequency discrimination will be presented [Fonseca, P.J., Münch, D., and Hennig, R.M., "How cicadas interpret acoustic signals," Nature 405,297–298 (2000)].

4:45

4pAB8. Bayesian model-based technique for termites detection. Asif Mehmood (Dept. of Elec. Eng., Univ. of Mississippi, University Mississippi), Orwa Tahaineh, and John Seiner (Univ. of Mississippi, University, MS)

This paper presents a model-based approach to detect termites from their head banging acoustic signals, and is derived from Bayesian probability theory. The termite head banging is the loudest and most diagnostic sound that termites make, and can be utilized for termite detection. The laser Doppler vibrometry system is used to obtain the termite head-banging signals from infested wood. An algorithm based on Bayesian probability theory is developed to detect termites' presence. The atomic model that represents the termites' data is the sum of decaying sinusoidal signals. First the model selection is performed that tells us about the number of vibration frequency components present in the data under observation. Once the correct model is known, then the vibration frequency that corresponds to termites' head banging frequency is determined. The calculations are performed using the Markov chain Monte Carlo method. Monte Carlo integration is then used to approximate the marginal posterior probabilities for all the parameters, including the number of exponentials and whether a constant offset is present. The performance of this algorithm is evaluated by testing it on experimental data, and the results obtained reveal the excellent performance of the algorithm.

Session 4pBB

Biomedical Ultrasound/Bioresponse to Vibration: Cardiovascular Applications of Ultrasound Contrast Agents

John S. Allen, Chair
Dept. of Mechanical Engineering., Univ. of Hawaii, Honolulu, HI 96822

Chair's Introduction—1:15

Invited Papers

1:20

4pBB1. A new high frequency microultrasound system with applications in cardiovascular research. F. Stuart Foster (Sunnybrook Health Sci. Ctr., Univ. of Toronto, 2075 Bayview Ave., and VisualSonics Inc., 3080 Yonge St., Toronto, Canada)

The development of preclinical imaging using micro-MR, CT, PET, SPECT, optical, and ultrasound technologies has created a new paradigm for imaging in the laboratories of biomedical researchers. Once considered a luxury for isolated multiuser centers, microimaging platforms are now becoming mainstream in bioresearch where quantitative *in vivo* imaging measurements of biomarkers and other endpoints are becoming a requirement of these investigations. Microultrasound has come a long way from its inception in the mid-1990s. This paper will detail the progression of preclinical microultrasound from mechanical to array based imaging systems. The technology of high frequency array based ultrasound imaging will be reviewed including details on the transducers and beamformer used in the first commercially available system. Applications of this system in the areas of cancer and cardiovascular disease will be described. The development of high frequency microbubble contrast modes based on linear and nonlinear signal processing will be discussed with relevant examples including imaging of VEGFR-2 and CD31 expression in disease models. All experiments with animals were done under a protocol approved by the Sunnybrook or VisualSonics Animal Care Committees. [The author declares a significant financial interest in VisualSonics Inc.]

1:40

4pBB2. Identifying and controlling acoustic bioeffects. Robyn K. Schlicher (Dept. of Chem. and Biomolecular Eng., Georgia Inst. of Technol., 315 Ferst Dr., Atlanta, GA 30332, rschlicher@chbe.gatech.edu), Joshua D. Hutcheson, Daniel M. Hallow (Georgia Inst. of Technol., Atlanta, GA 30332), and Mark R. Prausnitz (Georgia Inst. of Technol., Atlanta, GA 30332)

Ultrasound exposure causes internalization of a wide variety of molecules in cells and tissues, especially in the presence of acoustic cavitation. However, high levels of uptake are often accompanied by cell death. This study first analyzed intracellular uptake and cell viability after exposure of porcine carotid arteries to ultrasound *ex vivo* by confocal microscopy and found that at moderate ultrasound pressure, there was extensive uptake by endothelial cells with little uptake by underlying smooth muscle cells, whereas at high ultrasound pressure there was extensive endothelial cell death and increase uptake by smooth muscle cells. To understand the mechanisms involved in cell uptake and death, sonicated DU145 cells were analyzed by high level microscopy, flow cytometry, and chemical analyses. This work showed that uptake was caused by transient wounds created in the cell membrane that resealed within minutes after sonication. It also identified and quantified seven different cellular responses to this wounding, including cell repair modes and four different death modes. To save cells from apoptotic death, which was found to be mediated by calcium, cells were exposed to a calcium chelator, which rescued approximately half of the apoptotic cells from death.

2:00

4pBB3. A predictive model using myocardial contrast echocardiography for patients presenting to the emergency department with chest pain and a nondiagnostic electrocardiogram. Sanjiv Kaul (Cardiovascular Div., Oregon Health and Sci. Univ., Portland, OR 97201)

Risk stratification of patients presenting to an emergency department (ED) with suspected cardiac chest pain (CP) and an undifferentiated electrocardiogram (ECG) is difficult. We hypothesized that a risk score incorporating clinical, ECG, Rand contrast echocardiography variables [regional function (RF) and myocardial perfusion (MP)] obtained at the bedside would accurately predict adverse events in occurring within 48 h of ED presentation. A logistic risk model was developed in the initial 1166 patients (cohort 1), and validated in another 720 patients (cohort 2). Any abnormality or ST-T changes on ECG (OR 2.5, 95% CI:1.4–4.5, $p=0.002$, and OR 2.9, 95% CI:1.7–4.8, $p=0.001$, respectively), abnormal RF with normal MP (OR 3.5, 95% CI:1.8–6.5, $p=0.001$), and abnormal RF with abnormal MP (OR 9.6, 95% CI:5.8–16.0, $p=0.001$) were found to be significant multivariate predictors of nonfatal myocardial infarction or cardiac death. The estimate of the probability of concordance for the risk model was 0.82 for cohort 1. Likewise, in cohort 2, the c-index for the risk model was 0.83. In conclusion, a model based on variables obtained at the patient's bedside can be used to accurately risk stratify patients presenting to the ED with suspected cardiac CP and a nondiagnostic ECG. Its application could enhance care of CP patients in the ED.

2:20

4pBB4. Diagnostic ultrasound combined with targeted microbubbles improves recovery following acute coronary thrombosis.
Evan Unger, Terry Matsunaga (Dept. of Radiology, Univ. of Arizona, 1501 N. Campbell Ave., Tucson, AZ 85724), Feng Xie, and Thomas Porter (Univ. of Nebraska Medical Ctr., Omaha, NE 68198)

Forty-five pigs with acute coronary artery occlusions—low MI pulse sequence (CPS) to guide high MI (1.9 MI) pulses during infusion of platelet-targeted MBs or non-targeted MBs. Third group received no ultrasound/MB. All groups received pro-urokinase, heparin, and aspirin. Angiographic recanalization rates, resolution of ST elevation, and wall thickening were analyzed. Pigs receiving MB had more rapid replenishment of risk area (RA) (80% versus 40% for MB; $p=0.03$) and higher epicardial recanalization rates (53% versus 7% for pro-urokinase alone; $p=0.01$). Replenishment of contrast within RA with MB showed higher recanalization rates and higher rates of ST resolution (82% versus 21% for pro-urokinase alone; $p=0.006$). ST resolution occurred in six pigs (40%) with MB who did not have epicardial recanalization; five had recovery of wall thickening. Conclusions: IV MB with brief high MI DUS guided by CPS improves both epicardial recanalization rates and microvascular recovery.

2:40

4pBB5. Delivery of drug, or gene, to blood vessel wall using intravascular ultrasound and microbubbles. John Hossack (Biomed. Eng., Univ. of Virginia, Charlottesville, VA 22908-0759)

Atherosclerotic arteries are routinely treated using balloon angioplasty followed by stent placement. We developed a concept for a new therapy to prevent restenosis involving using ultrasound and microbubble-based drug/gene carriers. A number of early *in vitro* and *in vivo* results are presented. In particular, data for delivery of antiproliferative gene via microbubbles ruptured via catheter-based intravascular ultrasound (IVUS) at the site of vessel injury *in vivo* in a pig coronary artery are presented. Optimal design of a modified IVUS catheter is discussed. The proposed catheter includes a bubble port, an elongated single element transducer to provide radiation force to cause the bubbles to traverse to the vessel wall, and a bubble rupture transducer. The bubble rupture transducer is ideally coincident with an imaging IVUS scanning single element or annular phased array. These requirements provide the impetus to develop enhanced designs of both PZT based transducers and silicon micromachined transducers. In our initial pig study, transfection efficiency was measured using fluorescence microscopy and quantified as the percent of vessel perimeter cells expressing red fluorescent protein. We observed $23.3\pm6.0\%$ transfection in the treated vessel whereas the control exhibited $3.6\pm2\%$ transfection.

3:00—3:20 Break

3:20

4pBB6. Direct numerical simulations of bubble collapse near a tissue surface with the ghost fluid method. Hiroyuki Takahira and Kazumichi Kobayashi (Dept. of Mech. Eng., Osaka Prefecture Univ., 1-1 Gakuencho, Naka-ku, Sakai, Osaka 599-8531, Japan, takahira @me.osakafu-u.ac.jp)

The collapse of an air bubble induced by the interaction of an incident shock wave with the bubble near a gelatin or bone surface is investigated by using an improved Ghost Fluid Method (GFM). The motions of three phases for air inside the bubble, water, and gelatin (or bone) are solved directly by coupling the GFM with the level set method. The results show that the strong shock waves are generated not only when the bubble rebounds but also when the liquid-jet impacts the downstream surface of the bubble; the shock waves result in the depression of the gelatin (or bone) surface. Also, the penetration of the bubble into the depression of the gelatin surface is simulated successfully, which is in qualitative agreement with the experiment by Kodama and Takayama [Ultrasound in Med. & Biol. **24**, 723–738 (1998)]. It is also shown that the impulsive pressure at the bone surface caused by the bubble collapse is higher than that at the gelatin surface because the bubble collapse is accelerated by the high-pressure field generated by the reflection of the incident shock wave at the bone surface.

3:40

4pBB7. Acoustic characterization of echogenic liposomes: Attenuation and quantitative backscatter. Jonathan A. Kopechek (Dept. of Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, MSB 6163, Cincinnati, OH 45267, kopechja@uc.edu), Tyrone M. Porter (Boston Univ., Boston, MA), Constantin-C. Coussios (Univ. of Oxford, Oxford, UK), Stephen R. Perrin, Jr. (Univ. of Cincinnati, Cincinnati, OH), Shaoling Huang, David D. McPherson (Univ. of Texas Health Sci. Ctr., Houston, TX), and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH)

Echogenic liposomes (ELIPs) are being developed for use as ultrasonic contrast agents and as drug carriers for ultrasound-targeted drug delivery. Physical and acoustical characterization of ELIPs is necessary in order to determine the optimum parameters for diagnostic and therapeutic applications. In this study, ELIP samples at concentrations of 10, 20, and 50 μg/ml were exposed to ultrasound in pulse-echo mode at center frequencies of 2.25, 3.5, 7.5, 10, 15, and 30 MHz. The received echoes were analyzed to determine the attenuation and backscatter coefficients and the results were compared to a theoretical computational model. The sample chamber contained two 50-μm tungsten wires as reference scatterers. The echoes from the wires were acquired before and after addition of ELIP to determine the attenuation coefficient. The backscatter coefficient was determined by averaging the square of the RF amplitude between the wires and accounting for transducer parameters. Each transducer was calibrated and characterized in deionized water using PVDF hydrophones. The peak attenuation coefficient occurred at 7.5 MHz while the backscatter coefficient increased with frequency. This was in agreement with the computational model. These results provide important information for determining the optimum acoustical parameters for ELIP exposure in diagnostic and therapeutic applications. [Work supported by NIH 2RO1 HL059586-04A2 and ASA Hunt Postdoctoral Research Fellowship.]

4:00

4pBB8. Delivery of targeted echogenic liposomes in an *ex vivo* mouse aorta model. Kathryn E. Hitchcock, Jonathan T. Sutton (Dept. of Biomed. Eng., Univ. of Cincinnati, 231 Albert Sabin Way, MSB 6163, Cincinnati, OH 45267, hitchcke@email.uc.edu), Danielle N. Caudell, Gail J. Pyne-Geithman, D. Phil. (Univ. of Cincinnati, Cincinnati, OH), Melvin E. Klegerman, Shaoling L. Huang, Deborah Vela, David D. McPherson (Univ. of Texas Health Sci. Ctr., Houston, TX), and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH)

Optimal ultrasound parameters to enhance delivery of therapeutic-loaded echogenic immunoliposomes (ELIP) into the arterial wall are being developed for the treatment of atherosclerosis. The aim of this work was to determine whether anti-ICAM-targeted, rhodamine-labeled ELIP (Rh-ELIP) would adhere to and penetrate the vascular endothelium in atheromatous murine arterial segments with intravascular flow treated with 1-MHz continuous wave ultrasound (CW US). A broadband focused hydrophone, confocally aligned with the artery and 1-MHz transducer field was used as a passive cavitation detector (PCD). Arteries were insonified with 1-MHz CW US (0.49 MPa peak-to-peak pressure), and the PCD was used to verify the duration of the resulting stable cavitation. Perivascular saline was collected and analyzed spectrofluorometrically for the presence of Rh-ELIP. Arteries were prepared for histological analysis by a pathologist blinded to the exposure conditions. Arteries exposed to Rh-labeled ELIP and 1-MHz US exhibited greater adherence of Rh-ELIP to the vascular endothelium and greater passage of Rh-ELIP across the vessel wall. No damage was detected in any of the arteries on histology. These studies will aid in the development of a strategy for improving atheroma treatment without causing ultrasound-related tissue damage.

4:15

4pBB9. 120 kilohertz ultrasound-enhanced thrombolysis in a porcine intracerebral hemorrhage model. Azzdine Y. Ammi (Dept. of Biomedical Eng., Colleges of Medicine and Eng., Univ. of Cincinnati, 231 Albert Sabin Way, Cincinnati, OH 45267, ammia@ohsu.edu), Saurabh Datta (Siemens, CA), Stephen R. Perrin, Jr, Shauna L. Beiler, Christian R. Beiler, Kenneth R. Wagner, and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH 45267)

Ultrasound acts synergistically with thrombolytic agents, such as recombinant tissue plasminogen activator (rt-PA), to accelerate thrombolysis. The aim of the study was to demonstrate the efficacy of 120-kHz ultrasound-enhanced rt-PA thrombolysis in a porcine hemorrhagic stroke model *in vivo*. Clots were formed by infusing 3 ml of autologous blood into the frontal white matter of 30 mixed-bred Yorkshire pigs (20.5–3.1 kg) and incubated for 3 h. For these nonsurvival studies, six pigs received rt-PA alone (0.3 cc of 0.107 mg/ml), six received ultrasound alone, six received rt-PA plus ultrasound, six were sham-exposed (saline only), and six were controls (no ultrasound or rt-PA treatment). The clots receiving ultrasound treatment were insonified with a peak-to-peak pressure of 0.48 MPa *in situ* (80% duty cycle, and PRF of 1.7 kHz) for 30 min. Clots treated with rt-PA alone exhibited a volume loss of 55.0% and clots treated with rt-PA and 120-kHz ultrasound had a significantly higher volume loss of 75.2% and a higher penetration of rt-PA. Thus, 120-kHz pulsed ultrasound enhancement of thrombolysis has been demonstrated both *in vitro* and in an *in vivo* porcine hemorrhagic stroke model.

4:30

4pBB10. Modelling of oscillations of a microbubble in an elastic vessel. Sergey Martynov, Eleanor Stride, and Nader Saffari (Univ. College London, Torrington Pl. London WC1E 7JE, UK, s.martynov@ucl.ac.uk)

Encapsulated microbubbles have been extensively investigated as contrast agents for diagnostic ultrasound imaging and more recently for therapeutic applications such as drug delivery. However, theoretical models for microbubble dynamics exist either for encapsulated bubbles in an infinite volume of liquid, or for unencapsulated bubbles in a confined volume. In the present study, a finite-element method is applied to quantify the effects of both encapsulation and confinement in a blood vessel upon a microbubble's response to ultrasound. The effect of encapsulation is examined for polymeric and surfactant coatings. Elastic deformations of the vessel wall are described using a lumped-parameter model, treating the wall as a thin membrane. It will be shown that even at low acoustic pressures (10 kPa), the bubble oscillations can be significantly modified as a result of confinement. In particular, the frequency spectrum of the oscillations of a confined bubble is characterized by two modes. For relatively soft vessels, a high-frequency mode dominates, with the eigenfrequency increasing with the vessel stiffness. The eigenfrequency of the low-frequency mode decreases with the vessel length. The results will be discussed in the context of diagnostic and therapeutic applications.

4:45

4pBB11. Electrocardiogram-gated imaging of a mouse heart using a high-frequency annular array. Jeffrey A. Ketterling, Jonathan Mamou (Riverside Res. Inst., Lizzi Ctr. for Biomedical Eng., 156 William St., New York, NY 10038), and Orlando Aristizábal (Skirball Inst. of Biomolecular Medicine and New York Univ. School of Medicine, New York, NY 10016)

Gated imaging is a technique in which M-mode data, referenced to a feature of an ECG signal, are acquired at a series of lateral positions and then reassembled into B-mode images to achieve high effective frame rates. It is only effective for imaging objects with periodic motion. Annular arrays and synthetic focusing allow for an improved depth of field (DOF) and resolution versus single-element transducers. Here, a five-element, 35-MHz annular array was utilized in combination with ECG and respiratory gating to image adult mouse hearts. An experimental system was assembled to permit appropriate triggering conditions and to collect all 25 transmit-to-receive echo data sets from the annular array at a series of lateral positions. The system was initially tested with a hexagon-shaped target that rotated at 10 revs/s and generated a trigger each rotation. After system testing, data were acquired from an adult mouse heart using a trigger in phase with the R-wave of the ECG signal. The synthetically-focused B-mode images showed several cardiac cycles over a 1200 ms duration captured at 100 fps over a 1-cm depth and a 6-mm width. Data from the mouse heart were acquired with conventional monocycle excitation and also with coded excitation.

5:00

4pBB12. High-frequency imaging with targeted ultrasound contrast agents under vascular flow conditions. Pavlos Anastasiadis (Dept. of Mech. Eng., Univ. of Hawaii at Manoa, 2540 Dole St., Honolulu, HI 96822, pavlos@hawaii.edu) and John S. Allen (Univ. of Hawaii at Manoa, Honolulu, HI 96822)

Targeted ultrasound contrast agents (UCA) were conjugated with biotinylated ligands, which allow them to adhere to specific diseased sites of interest. The overall adhesion of contrast agents is influenced by both the flow mediated gene expression and the local hydrodynamic forces acting on the contrast agents' trajectories. The ability to bind under pulsatile flow conditions and scatter sound at high frequencies is important for potential applications involving intravascular ultrasound. Human aortic endothelial cells (HAECs) in a flow chamber were exposed to a steady shear pretreatment over 12 h using a peristaltic pump system and subsequent pulsatile waveforms. To stimulate an inflammatory reaction in the HAECs, cells were subsequently exposed to rhTNF945; (Roche Pharmaceuticals). Different flow rates were applied, and the binding efficacy of targeted UCA was evaluated. Real-time transendothelial electrical impedance measurements and simultaneous acoustic measurements were performed on the same specimen with a scanning acoustic microscope. Applications related to targeting plaque in blood vessels are discussed. [This work was supported by the National Institutes of Health Grants NIH 2 P20 RR016453-05A1 and NIH 2 G12 RR0030161-21.]

Session 4pNSa

Noise, Architectural Acoustics, and ASA Committee on Standards: Soundscape Techniques and Applications—Community and Urban Environments

Brigitte Schulte-Fortkamp, Cochair
Inst. of Fluid Mechanics and Engineering, Technical Univ. Berlin, 10587 Berlin, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 30 Lafayette Sq., Suite 103, Vernon, CT 06066

Chair's Introduction—1:00

Invited Papers

1:05

4pNSa1. Soundscape approach in progress—Report on the current stage. Brigitte Schulte-Fortkamp (TU-Berlin, Inst. of Fluid Mech. and Eng. Acoust., Einsteinufer 25, D-10587 Berlin, Germany)

Soundscapes have become an important issue in environmental acoustics. Following Soundscape workshops in Vancouver, Salt Lake City, and Miami, a Soundscape Symposium took place in Berlin, Germany, and brought together expertise from all over the world to work on concepts, approaches, analyses, applications, as well as source- and pattern recognition with respect to Soundscapes. Therefore, acousticians, architects, city planners, engineers, psychologists, sociologists, and the people concerned were invited. The aim of the workshop was to define Soundscapes for future work. Moreover, concepts, approaches, analyses, and applications were related to the categorization of Soundscapes as urban, cultural, and wilderness, also focussing on source and pattern recognition. This paper will report on the results to outline the next steps towards standardization.

1:25

4pNSa2. Integrating soundscape analysis into the National Environmental Policy Act process: A case study. George Luz (Luz Social and Environ. Assoc., 4910 Crowson Ave, Baltimore, MD 21212, Luz-Assoc.@msn.com)

A challenge for the proponents of soundscape analysis within the U.S. is how to integrate subjective observations with the quantitative requirements of the National Environmental Policy Act (NEPA), which, historically, emphasize physical measurements of all pollutants, including sound pollution. This paper describes a case study in which an attempt is made to integrate the subjective impressions of a soundscape analysis with objective measurements of equivalent sound level (LEQ). The occasion for this effort was a Base Realignment and Closure (BRAC) action in which the proposed reuse for a former military installation is park and recreational use. The quantitative framework for this study was a graphic used by the U.S. Army Environmental Hygiene Agency (USAEHA) during the 1980s in which the 24 h pattern of outdoor noise exposure is displayed as the minimum, average, and maximum values of 10 min LEQ over the course of a week of measurements. Although the intent of the USAEHA approach had been to determine conformance of military residential areas to Army guidelines, their method for displaying measurement data proved to be amenable to soundscape analysis as well.

1:45

4pNSa3. Soundscaping the aesthetic—Beyond measurement and assessment. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Lowell, MA 01854, Alex_Case@uml.edu)

Soundscape in the context of community noise alone risks missing the important subjective reactions people have to the aesthetic value of any sound environment. The music recording and film sound industries have been recording sounds for art and entertainment, without regard for objective assessment, for more than a hundred years. Soundscape design has the opportunity to borrow from the time-tested recording, signal processing, and playback techniques developed by recording engineers where appropriate, learning from their search for emotional, intellectual, and uniquely human reactions. An introduction to nonobjective recording techniques is presented, with emphasis on some of the more counter-intuitive parts of the recording craft.

Contributed Paper

2:05

4pNSa4. A study on the modified urban soundscape of a city due to introduction of elevated structures. Kalaiselvi Ramasamy and Ramachandraiah Alur (Dept. of Civil Eng., IIT-Madras, India)

The urban soundscape of cities in a developing country like India are slightly varied by virtue of the fact that the composition of the traffic is heterogeneous accompanied by variance in road geometry and varying density of the buildings on the either side of the road and other community noise sources. Urban planning plays a vital role in organizing a city's traffic flow. To avoid congestion of traffic streams introduction of flyovers in the traffic flow happens to be a common feature in many urban environments. Through introduction of flyovers an increase or decrease in environmental noise characteristics occurs. In this paper a noise mapping study has been attempted along with field measurements of L10, L50, L90, and Leq to understand the soundscape of the city due to such types of modified topography. It also describes how the local characteristics of the city and changed topography alters the soundscape of the city. The theoretical computation of noise levels has been carried out using the sound plan software. It is observed that a reasonable reduction of Leq occurs in the immediate vicinity of noise sensitive areas of significant buildings such as hospitals and educational campuses.

THURSDAY AFTERNOON, 21 MAY 2009 EXECUTIVE SALON II/III, 2:30 TO 5:10 P.M.

Session 4pNSb

Noise, Architectural Acoustics, and ASA Committee on Standards: Soundscape Techniques and Applications—Wilderness and Park Soundscapes

Nancy S. Timmerman, Cochair
Timmerman Consultant in Acoustics, 25 Upton St., Boston, MA 02118-1609

Paul D. Schomer, Cochair
Schomer & Associates Inc., 2117 Robert Dr., Champaign, IL 61821

Chair's Introduction—2:30

Invited Papers

2:35

4pNSb1. Overview of on-going Federal Aviation Administration and National Park Service collaborative research efforts in support of the National Parks Air Tour Management Act. Cynthia Lee (U.S. Dept. of Transportation, Res. and Innovative Technol. Admin., Volpe Ctr., 55 Broadway, RVT-41, Cambridge, MA 02142, Cynthia.Lee@dot.gov)

In support of the National Parks Air Tour Management Act of 2000 (NPATMA), the FAA and NPS are developing air tour management plans (ATMPs) for approximately 100 national parks. ATMP objectives are to develop acceptable and effective measures to mitigate or prevent significant adverse impacts, if any, of commercial air tour operations upon the natural and cultural resources of and visitor experiences in national parks and abutting tribal lands. In accordance with NPATMA, any methodology adopted by a federal agency to assess air tour noise under this Act shall be based on reasonable scientific methods. Both agencies acknowledge that additional research is needed. This paper presents an overview of Volpe Center contributions to the collaborative effort by the FAA and NPS to develop improved methods to (1) quantify park soundscapes (natural and non-natural); (2) enhance computer modeling capabilities through aircraft source noise database expansion and advanced research into the factors which affect aviation noise propagation for complicated environments, such as national parks; and (3) assess the effects of aircraft overflights on park visitor experience, including the metrics used in these assessments. Improved methods developed in these efforts will also be used to support other agency projects related to aviation noise.

2:55

4pNSb2. Visitor perception of park soundscapes: An approach and research plan. Paul Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssoc..com) and G. Randy Stanley (Natl. Park Service, Fort Collins, CO 80525)

In the United States, much of the research on the soundscape in national parks and wilderness areas has centered on so-called dose-response relations with the aim of stating: How much manmade noise is unacceptable? Based on their experience in urban areas, this approach has been advocated primarily by the aviation noise community. But there is another school of thought. Visitors to national parks are guests and patrons of what the park has to offer. Just as juries of listeners are used to judge the sound quality of automobiles and home-appliances, juries of listeners could also be used to judge the sound quality in national parks. The current plan is to develop outdoor sound quality measurement and prediction methods and standards by which the U.S. National Park Service can accomplish their assessments of the acoustic ambient. This paper discusses the plan for this measurement protocol development and indicates some standards to be developed.

3:15

4pNSb3. "The Grand Canyon" versus "Soundscape of Nowhere (continued)". Dickson J. Hingson (Sierra Club—Natl. Parks and Monuments Committee)

More than 21 years have elapsed since the National Parks Overflights Act mandated the prompt "substantial restoration" of the natural quiet of the aircraft-noise imperiled soundscape of the Grand Canyon National Park. However, as of 2008, long-anticipated, critical compliance benchmarks have still not been timely met in the Park. The past two Administrations have not conformed to specifications/standards/deadlines set or appropriate to the NPS under its legal mandates. However, every battle has its turning point. *Will 2009 be the turning point to a quiet Canyon?* Success will require immediate NPS application of long-established restoration standards (based on "audibility"), Park zoning considerations, and buttressing with emerging supplemental noise indicators, which trigger loudness and temporal impulsiveness mitigations. Effectiveness of imminently anticipated management actions in the form of a soon forthcoming 2009 environmental impact statement and stepped up political oversight will be examined. These will pit restoration of the authentic Grand Canyon wilderness soundscape against the current, unsavory option: "the Soundscape of Nowhere." The protracted Grand Canyon imbroglio illuminates similarly unmet, pressing restoration needs, along with the need for increased executive /congressional oversight, re low-altitude air tour noise unacceptably continuing at similarly impacted, famed national parks, which otherwise remain subject to long-term, aviation noise impairment.

3:35

4pNSb4. A case for the importance of context for soundscape research in parks and protected areas. G. (Randy) Stanley (Natural Sounds Prog., U.S. Natl. Park Serv., 1201 Oakridge Dr., Ste. 100, Ft. Collins, CO 80525, Randy_Stanley@nps.gov) and Paul Schomer (Schomer and Assoc., Inc., Champaign, IL 61821)

Over the past decade or so, there have been a number of research studies, summarized herein, that indicate context is a potentially important intervening variable in assessing the soundscape in a park or wilderness area. Context is primarily provided by the corresponding visual landscape but may, in testing situations, be provided in other ways such as a verbal or written description. Context has been shown to be important in sound quality testing of automobile and product sounds, but its importance in assessing environmental sounds is less well documented. This paper discusses these various issues and suggests possible courses of action.

Contributed Papers

3:55

4pNSb5. The acoustical status of U. S. national parks. Kurt Fristrup (NPS Natural Sounds Program, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO, 80525)

Acoustical monitoring data have been collected for more than 40 national park units, spanning a wide range of regions, resources, and park purposes and values. Comparative analyses of these data reveal the extent of noise pollution in national parks. Transportation noise is audible in many wilderness areas more than 30% of the daylight hours, with peak hourly levels approaching 70%. Peak levels from noise events can be more than 50 dB above the natural background under air tour routes, and between 10–20 dB above background for high altitude aircraft. Hourly Leq values can be as much as 6 dB above natural levels. These data also reveal the degree to which various noise metrics covary in these natural settings. This information helps inform selection of a compact set of metrics that can assess several functional impacts of noise while minimizing redundancy.

4:10

4pNSb6. Baseline sound monitoring plan for Grant-Kohrs Ranch national historic site. Robert C. Maher (Elec. & Comput. Engr., Montana State Univ., 610 Cobleigh Hall, Bozeman, MT 59717-3780, rob.maher @montana.edu)

Grant-Kohrs Ranch National Historic Site (GRKO), located just north of Deer Lodge, Montana, is a working cattle ranch commemorating the heritage of cowboys and stock growers in the history of the American West during the 19th and 20th centuries. The U.S. National Park Service (NPS) maintains the site according to its charter as a working ranch, with all the sights, sounds, and sensations associated with ranching. The cultural soundscape of the working ranch is considered essential to visitor enjoyment and understanding. Several anticipated changes in the neighboring community of Deer Lodge may affect the visitor experience at GRKO, including anticipated expansion of the local airport, increasing interstate highway traffic, and proposals to establish a rifle shooting range nearby. Because GRKO currently has no data characterizing the natural and cultural sounds of the park, this project was commissioned to monitor and evaluate the natural, cultural, and community sounds that comprise the ambient acoustic environment of the historic site over the period of one calendar year. The baseline acoustical data are analyzed and documented in a format suitable for management purposes by the NPS (the sponsor of this study), and by the NPS Natural Sounds Program Office.

4:25—5:10 Panel Discussion

Session 4pPA

Physical Acoustics, Underwater Acoustics, and Engineering Acoustics: A Half-Century with the Parametric Array II

Kenneth G. Foote, Cochair
Woods Hole Oceanographic Inst., Woods Hole, MA 02543

Murray S. Korman, Cochair
Physics Dept., U. S. Naval Academy, Annapolis, MD 21402

Contributed Papers

1:20

4pPA1. Evaluation of parametric array technology for acoustic landmine detection. Murray S. Korman (Dept. of Phys., U.S. Naval Acad., Annapolis, MD 21402, korman@usna.edu), Antal A. Sarkady and Frederick R. Tolle (U.S. Naval Acad., Annapolis, MD 21402)

There has been interest in using the parametric array for obtaining a highly directional low frequency source in acoustic landmine detection [M. S. Korman, J. Acoust. Soc. Am. **122** (2007)]. Earlier experiments used a 0.5 m commercial parametric array made up of 70 elements located 1.5 m directly over the target. The array was driven by a 110–1100 Hz swept sine audio modulated 65 kHz tone. A VS 1.6 inert plastic antitank landmine was buried 2.5 cm deep in dry sifted masonry sand in a concrete box. The laser Doppler vibrometer to microphone rms response was sufficient to measure the "on target" to "off target" contrast ratios of 20 and 3 for peaks near 850 and 1050 Hz, respectively (upon signal averaging) but missed the largest peak near 150 Hz (where the SPL was 40 dB per 20 μPa) among others. Recent "forward looking" experiments aligned the array beam axis at 30° from grazing. Sweeping from 110–210 Hz (1600 points) at 16 s/sweep and signal averaging 20 sweeps produced a contrast ratio of 10 for resonance at 148 Hz. Considerable improvement in SPL is necessary in order to make this technology practical for low frequency applications. [Work supported by ARL.]

1:35

4pPA2. Spatial phase-inversion technique for parametric source with suppressed carrier. Tomoo Kamakura, Hideyuki Nomura, and Shinichi Sakai (Dept. of Electronis, Univ. of Electro-Commun.s, 1-5-1, Chofugaoka, Chofu-shi, Tokyo 182-8585, Japan)

Two planar projectors with the identical rectangular apertures are placed side by side. Both the projectors are radiating bifrequency ultrasound beams of finite amplitude in the air. The frequencies are 40 and 42 kHz but the initial phases are different. Especially, two extreme cases are considered: one is conventional in-phase driving, and the other is phase inversion driving. Sound pressure profiles were measured along and across the sound beam axis for the primary waves and the difference frequency wave of 2 kHz. The second and third harmonic components of the difference frequency were measured as well. Obviously, the pressure levels of the primary waves were suppressed considerably near the beam axis due to phase cancellation when the driving signals were out-of-phase by 180 degrees. The beam pattern of the difference frequency was, however, almost the same as the case where the signals were in phase. Interestingly, the pressure levels of the harmonics were reduced more than ten decibels. The validity of experimental results has been supported by good agreement with the theoretical predictions based on the Khokhlov-Zabolotskaya-Kuznetsov model equation. [Work supported by JSPS.]

1:50

4pPA3. Infrasound-convection nonlinear interaction. Konstantin Naugolnykh (Earth System Res. Lab., NOAA/Univ. of Colorado/Zeltech, Boulder, CO 80305)

The temperature stratified atmospheric layer is unstable and convection flow can be developed in such an area. Convection happens because warm less dense air goes up while cooler air comes down. The presence of infrasound produces modulation of convection flow and the sound wave amplification. The process of infrasound-convection nonlinear interaction is considered in the present paper. The equations of a compressible fluid convection are derived, which is characterized by the modified Rayleigh number, in comparison to this number for the incompressible flow, and condition of infrasound amplification is obtained. The increment of infrasound amplification turned out to be proportional to the Rayleigh number.

2:05

4pPA4. Sonar material acoustic property measurements using a parametric array. Victor F. Humphrey (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, SO17 1BJ, UK, vh@isvr.soton.ac.uk.), Stephen P. Robinson, Graham A. Beamiss, Gary Hayman (Natl. Physical Lab., Teddington, Middlesex, TW11 0LW, UK), John D. Smith (DSTL, Porton Down, Salisbury, Wiltshire, SP4 0JQ, UK), Michael J. Martin (QinetiQ Ltd., Farnborough, Hampshire GU14 0LX, UK), and Nicholas L. Carroll (QinetiQ Ltd., Park, Winfrith Newburgh, Dorchester, Dorset, DT2 8XJ UK)

The use of a parametric array as an acoustic source for measuring the acoustic properties of materials for use in sonar systems is considered. Techniques of measuring the wideband transmission and reflection properties of limited size test panels are described and the advantages of using a parametric array in terms of reducing edge diffraction errors are discussed. Results for a range of materials under ambient pressure over the range 10–200 kHz are presented to illustrate the potential of the technique. The implementation of the approach in a pressure vessel at the UK National Physical Laboratory is also described and illustrated with example results obtained over the frequency range 2–50 kHz for two test objects that have predictable behavior. The potential of the technique is also illustrated with experimental results for viscoelastic test panels for hydrostatic pressures up to 2.8 MPa.

2:20

4pPA5. Parametric array application for long range ocean sounding. Konstantin Naugolnykh (Earth System Res. Lab., NOAA/Univ. of Colorado /Zeltech, Boulder, CO 80305) and Igor Esipov (N. Andreev Acoust. Inst., 4 Schvernik St., Moscow, Russia)

The parametric array (PA) is a nonlinear transducer that generates narrow, sidelobe-free beams of low frequency sound, through the interaction of high frequency pump waves. PA was suggested by Peter J. Westervelt, winner of the Lord Rayleigh Medal, at the same time this devise invention was underway in the Soviet Union made by Zverev and Kalachev. Remarks of PA applications for long distance ocean sounding is presented in the present paper. Experimental test of a high-frequency PA was made on the Black Sea coast by Esipov *et al.*, and investigations of powerful PA characteristics was performed in a ocean by Andebura *et al.*, in 1990. Then large-scale ocean vortices sensing at a distance of 1000 km was realized by Esipov *et al.* [Acoust. Zh., **40**(1), 71–75 (1994)]. In both experiments, the transducer of R/V "Boris Konstantinov" was used with parameters: pump wave power

W=20 kW, pump wave frequency f=3 kHz, parametric signal frequency F=230–700 Hz. Obtained results indicate the efficiency of PA application for long distance ocean sounding. [Work supported by ISTC project 3770 and NATO Grant No. ESP.NR.NRCLG 982524.]

2:35

4pPA6. Observing Atlantic herring by parametric sonar. Olav Rune Godø (Inst. of Marine Res., PO Box 1870, N-5817 Bergen, Norway, olav.rune.godoe@imr.no), Kenneth G. Foote (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), Johnny Dybedal (Kongsberg Defence & Aerosp. AS, Stjoerdal, Norway), and Eirik Tenningen (Inst. of Marine Res., Bergen, Norway)

Atlantic herring (*Clupea harengus*) has been observed in situ by the Kongsberg TOPAS PS18 parametric sub-bottom profiling sonar, with nearly horizontal transmit and receive transducer arrays mounted flush on the hull of R/V G. O. Sars, at N71.4 E16.3. The primary frequencies are in the band 15–21 kHz, with nonlinearly generated difference frequencies in the band 0.5–6 kHz. The beamwidths are 6–10 deg depending on frequency and range, but with exceedingly low sidelobes. The observation of herring in schools and layers was accomplished with the vessel both at rest and sailing at its ordinary survey speed of 10 knots. The observations with the parametric sonar were confirmed by concurrent, synchronized observations with the Simrad EK60/38-kHz scientific echo sounder and by trawling with a pelagic net. The herring length varied from 28.0–36.5 cm. Present work suggests that parametric sonar will become a powerful new tool in marine ecosystem studies, enabling the numerical density of schooling and shoaling fish to be determined, and the size of swimbladder-bearing fish to be estimated by detection of swimbladder resonance. [Work partly supported by Norwegian Research Council Grant no. 184705.]

2:50

4pPA7. Range compensation function for echo integration in transducer near fields, with special reference to parametric sonar. Kenneth G. Foote (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, kfoote@whoi.edu)

Range compensation functions are applied to echo signals so that the corresponding range-compensated echo intensity equals the volume backscattering coefficient to within a multiplicative constant. For echo integration in a transducer far field, the transmit and receive pressure fields each decrease inversely with range r, and the well-known range compensation function, applied to intensity, is $r^2\times 10^{\alpha r/5}$, where α is the absorption coefficient. For echo integration in a transducer near field, the range compensation function is that of the far field modified by a multiplying factor. This factor is, approximately, the ratio of the square of the surface integral of the field intensity at r due to extrapolation from the far field, assuming inverse range dependence of the amplitude, to the product of the surface integrals of transmit and receive intensities at r. This general range compensation function is developed for the special case of echo integration with a parametric sonar in which scatterers are ensonified in a region where the nonlinearly generated difference-frequency field is growing and echoes are received in the far field of a collocated, linearly operating transducer. It is evaluated numerically for the Kongsberg TOPAS PS18 parametric subbottom profiling sonar, with primary-frequency band 15–21 kHz and difference-frequency band 0.5–6 kHz.

3:05

4pPA8. Standard-target calibration of a parametric sonar over the difference-frequency band, 1–6 kilohertz. Kenneth G. Foote (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543, kfoote@whoi.edu), Johnny Dybedal (Kongsberg Defence & Aerosp. AS, N-7501 Stjoerdal, Norway), and Eirik Tenningen (Inst. of Marine Res., N-5817 Bergen, Norway)

The Kongsberg TOPAS PS18 parametric sub-bottom profiling sonar operates over the frequency band 15–21 kHz, with nonlinearly generated difference-frequency radiation in the band 0.5–6 kHz. The TOPAS transducer mounted on R/V G. O. Sars is flush with the hull in the near-horizontal plane. The sonar has been calibrated by the standard-target method using a 280-mm diam sphere of aluminum alloy 6082 T6 [K. G. Foote *et al.*, J. Acoust. Soc. Am., **121**, 1482–1490 (2007)]. The target was suspended beneath the vessel at each of three ranges, successively 100, 200, and 300 m. Because of conditions in Soerfolla fjord on Dec. 10, 2008, the target sphere was moving slowly relative to the vessel. Its instantaneous position was determined by geometrical considerations through synchronous observation with the Simrad EK60/38-kHz scientific echo sounder, with split-beam transducer mounted approximate to the TOPAS transducer. Data were collected for a number of parameter settings for each of three signal types: continuous wave, chirp, and Ricker pulse. Measurements are compared with predictions based on laboratory measurements of the frequency-dependent sensitivities of the parametric transmitter and conventional linear receiver, using a range-compensation function based on theoretical nearfield modeling. [Work partly supported by Norwegian Research Council Grant No. 184705.]

3:20—3:40 Break

Invited Papers

3:40

4pPA9. Resonant mode conversion in superfluid 4 helium and in solids. Steven L. Garrett (Graduate Program in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804)

One component of the success of the Westervelt analysis of the parametric array was that sound propagation in most gases and liquids exhibit very little dispersion. Therefore, the pump waves and their nonlinearly generated product propagate at very nearly the same speed. When an acoustic medium can support more than one sound mode with different propagation speeds, then a parametric array can be created by the nonlinear interaction of the two slower sound waves to produce the faster wave, if the slow wave propagation directions are not collinear. Bulk superfluid helium, at temperatures below 2.1 K, will support a slower (thermal) sound mode and a faster (mechanical) sound mode. An experiment performed in 1977 will be described that used a waveguide to create thermal (temperature-entropy) sound waves while controlling their angle of intersection. A mechanical (pressure-density) wave was produced and its amplitude determined the thermodynamic coupling constant between density changes and the Galilean invariant square difference in the normal and superfluid particle velocities that characterize the thermal sound wave amplitude. An earlier experiment will also be described that demonstrated the nonlinear conversion of two shear waves to a longitudinal wave in aluminum. [Work supported by the Office of Naval Research.]

4:00

4pPA10. Quantum acoustics, the second law and the end fire array. Seth Putterman (Dept. of Phys., UCLA, Los Angeles, CA 90095) and Paul H. Roberts (UCLA, Los Angeles, CA 90095)

The fundamental nonlinear interaction which leads to the scattering of waves and to the end fire array yields a Boltzmann equation for sound, a one fluid theory of superfluid helium [S. Putterman, P. H. Roberts, Physics Letters, **89A**, 444 (1982)] and a route to the quantum theory of interacting phonons [M. Cabot, S. Putterman, Phys. Lett. **83A**, 91 (1981)]. Along with a review of these consequences of nonlinear classical acoustics the conundrum of the consistency of the second law of thermodynamics with the Hamiltonian nature of the end fire array will be exposed but not resolved [S. Putterman, P. H. Roberts, Phys. Rep., **168**, #4 (1988)].

Contributed Paper

4:20

4pPA11. Acoustic nonlinearity in fluorinert. Cristian Pantea, Dipen N. Sinha, Curtis F. Osterhoudt, and Paul C. Mombourquette (Los Alamos Natl. Lab., MPA-11, MS D429, Los Alamos, NM 87545, pantea@lanl.gov)

Fluorinert FC-43 nonlinearity was investigated using two approaches: (i) a finite amplitude method with harmonic production and (ii) a nonlinear frequency mixing in the fluid with consequent beam profile measurement of the difference frequency. The finite amplitude method provides information on the coefficient of nonlinearity, β, through the amplitudes of the fundamental and the second harmonic, at a certain transmitter-receiver distance. A calibrated hydrophone was used as a receiver in order to obtain direct pressure measurements of the acoustic waves in the fluid. The role of transmitter-receiver distance in β determination is investigated. In the second approach, a single transducer is used to provide two high-frequency beams. The collinear high-frequency beams mix nonlinearly in the fluid resulting in a difference frequency beam and higher order harmonics of the primaries. The difference frequency beam profile is investigated at lengths beyond the mixing distance. The experimental data are compared with the KZK theory.

Invited Paper

4:35

4pPA12. Flow through orifices caused by large-amplitude asymmetric sound waves: Theory and experiment. Peter J. Westervelt (Dept. of Phys., Brown Univ., Providence, RI 02912, abwpjw@cox.net)

When the displacement amplitude of an acoustic wave consisting of equal parts fundamental and second harmonic exceeds the diameter of a circular orifice, steady flow is generated whose direction is determined by the phase of the two wave components [P. J. Westervelt, Ph.D. thesis "The interaction of a finite amplitude acoustic wave with small obstacles and orifices," (MIT, 1951)]. Some applications to semipermeable biological membranes are discussed. [Work supported by the Office of Naval Research].

THURSDAY AFTERNOON, 21 MAY 2009 GRAND BALLROOM I, 1:00 TO 4:00 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Physiology; Aging; Hearing Loss (Poster Session)

Kathryn H. Arehart, Chair
Dept. of Speech, Language and Hearing Science, Univ. of Colorado at Boulder, Boulder, CO 80309

Contributed Papers

All posters will be on display from 1:00 p.m. to 4:00 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 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

4pPP1. Effect of stimulus onset delay on auditory cortex neural responses to voice pitch feedback perturbation. Roozbeh Behroozmand and Charles R. Larson (Speech Physiol. Lab., Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208)

It has previously been shown that the auditory neural responses to voice F0 feedback perturbation are suppressed during active vocalization compared to passive listening to the playback. However, a study in primates showed that the vocalization-induced suppression enhances auditory sensitivity to feedback perturbation. This evidence suggests that the cortical neural responses to self-vocalization reflect suppression to vocal onset and excitation in response to perturbations in voice pitch feedback. In this study, we investigated the effect of stimulus onset delay on cortical neural responses to voice F0 feedback perturbation. Event-related potentials (ERPs) were recorded in human subjects in response to simultaneous (0 ms) and delayed (500 ms) PSS (+200 cents) during active vocalization and passive listening conditions. Results showed that, for the delayed PSS, the P200 peak magnitude was larger during vocalization compared with passive listening. This finding suggests that vocalization enhances auditory cortex responsiveness to deviations in voice pitch feedback following the onset of vocalization. This enhancement might be due to changes in tuning properties of auditory neurons by the vocal motor system that helps to detect and correct for vocal errors during speech production.

4pPP2. Event-related potential correlates of online monitoring of auditory feedback during vocalization. Colin S. Hawco, Jeffery A. Jones, and Todd R. Ferretti (Dept of Psych., Wilfird Laurier Univ., 75 University Ave. West, Waterloo, ON N2L 3C5, Canada)

When speakers hear the fundamental frequency (F0) of their voice altered, they shift their F0 in the direction opposite to the perturbation. The neural mechanisms underlying this response are poorly understood. In the present study, event-related potentials (ERPs) were used to examine the neural mechanisms used to detect alterations in auditory feedback during an ongoing utterance. Participants vocalized for 3 s, and heard their auditory feedback shifted by 0, 25, 50, 100, or 200 cents for 100 ms midutterance. In two sessions, participants either vocalized at their habitual pitch, or matched a target pitch. A mismatch negativity (MMN) was observed, with the amplitude positively related to the size of the perturbations. No differences were found between sessions. The F0 compensation response was found to be smaller for 200 cent shifts than 100 cent shifts, and a positivity was observed in the ERPs for a 200 cent shift. This result suggests that a 200 cent shift may be perceived as externally (rather than internally) generated. The

presence of an MMN, and no earlier (N100) response suggests that the underlying sensory process used to identify and compensate for errors in midutterance may differ from feedback monitoring at utterance onset.

4pPP3. Ictal and interictal changes in central auditory processing. David B. Daly and David M. Daly (Box 210855, Dallas, TX 75211, dave.daly@stanfordalumni.org)

Disordered functioning manifest in seizures can also give rise to subtle, pervasive interictal changes. We have used sets of synthesized acoustic stimuli, concurrent EEG and AED blood-level monitoring to evaluate changes in a 24-year old female with focal seizures. During ictus patient becomes mute, she can hear but not comprehend, and may then be briefly amnestic. EEG revealed sharp/slow waves over left anterior temporal regions, with occasional bilateral discharges. Levels of two AED fluctuated 50%, with peaks 2 h apart. Performance on 4-min sets of auditory testing with ge-ye and be-de-ge fluctuated from well defined ($p<0.001$) near either peak AED, to near-chance levels; three-choice vowel sets were well defined throughout. Disruptions, which involved contralateral homologous areas as well as surrounding ipsilateral areas, are consistent with augmented inhibition.

4pPP4. Measures of wideband power reflectance in otosclerotic ears. Marcin Wróblewski (Graduate Ctr., City Univ. of New York and Dept. of Otolaryngol., NYU Langone Medical Ctr., 550 1st Ave., NBV 5E5, New York, NY 10016, marcin.wroblewski@nyumc.org), Arlene C. Neuman, Nancy Jiang, and Anil K. Lalwani (NYU Langone Medical Ctr., New York, NY 10016)

Wideband reflectance (WBR) is a new method of evaluating middle ear function. While several studies have reported WBR data obtained from persons with normal hearing and from children with otitis media, few data have been reported describing results typical of other types of middle ear pathology. The purpose of the present study was to collect data from a group of persons with clinically diagnosed otosclerosis and to compare the measures of WBR (including reactance, resistance, impedance magnitude, and transmittance) to measures obtained from persons with normal middle ear function. WBR data were collected from 17 preoperative and 18 postoperative otosclerotic ears as well as from 57 ears without middle ear pathology. Measures from four otosclerotic ears allowed for a direct comparison of pre- and postoperative middle ear function. There was a variety of responses from ears with otosclerosis. Preoperatively, while some otosclerotic ears showed a pattern reported in previous case studies (i.e., increased WBR below 1000 Hz indicating enlarged middle ear stiffness), others fell within the normal range. Postoperatively, some ears showed a pattern typical of an increased mass component of middle ear impedance (i.e., decreased WBR below 1000 Hz), while others fell within the normal range.

4pPP5. Comparison of measurements at ambient pressure on clinical immittance and wideband acoustic transfer function systems. Kim Schairer, Brooke Morrison, Ellyn Steininger, and Cynthia Fowler (Univ. of Wisconsin, 1975 Willow Dr., Rm. 373, Madison, WI 53706)

The purpose of the study was to compare acoustic admittance recorded using probe tones of 226, 678, and 1000 Hz on a clinical immittance system with admittance and reflectance recorded on a wideband acoustic transfer function (WATF) system. The WATF system uses a click probe, which yields measurements across a broad range of frequencies with one test, whereas multiple tests are required at individual frequencies on the clinical system. Thus, the WATF system has the potential to provide more information in a shorter amount of time. The hypothesis was that admittance would be comparable between the two systems, which would provide support for clinical use of the WATF system. Measurements were taken at ambient pressure (i.e., 0 Da Pa) in the ear canals of adults with normal hearing and middle-ear function. Because the clinical system reports only information at the tympanometric peak by default, the data at 0 Da Pa were manually recorded by entering the cursor mode. In general, the results at all three probe frequencies support the hypothesis, although the relationship was stronger for the two higher probe tone frequencies. The clinical implications and effect of analysis bandwidth on the WATF system will be discussed.

4pPP6. An electroacoustical analog for estimating sound pressure level at the tympanic membrane at high frequencies. Janice L. LoPresti (Knowles Electron.s LLC, 1151 Maplewood Dr. Itasca, IL 60143, janice.lopresti@knowles.com), Karrie Recker, and Tao Zhang (Starkey Lab., Inc., Eden Prairie, MN 55344)

For various applications, it is useful to know the sound pressure level (SPL) at the tympanic membrane (TM). However, measuring the SPL close to the TM is not clinically feasible, due to safety and discomfort concerns. Therefore, it is desirable to estimate the SPL at the TM using measurements away from the TM. This is challenging at high frequencies where the effect of canal geometry becomes significant as the wavelength of the acoustical source becomes comparable to the dimensions of the canal. In this study, an electroacoustical analog was used to estimate the SPL at the TM for ten participants. The analog is comprised of a transducer (i.e., sound source), occluded ear canal, and the middle ear [Pascal *et al.* (1998)]. The ear model was optimized for each individual using a 2-parameter fit using real ear measurements further away in the canal. The optimization did not require inputs that may be difficult to obtain such as the exact canal geometry. A simulated transfer function was created and applied to each measured real-ear response to generate the estimated response at the TM. The model was verified using the SPL measured at the TM.

4pPP7. Evidence for dynamic cochlear processing in otoacoustic emissions and behavior. Kyle P. Walsh, Edward G. Pasanen, and Dennis McFadden (Dept. of Psych., Ctr. for Perceptual Systems, Univ. of Texas at Austin, 1 University Station A8000, Austin, TX 78712)

Psychophysical research suggests that active cochlear processing may partially explain temporal effects observed in certain auditory masking tasks [Strickland (2001), (2004)]. To investigate this possibility, this study examined the relationship between subjects' psychophysical performance and the responses of their cochleas to the same stimulus waveforms. Stimulus-frequency otoacoustic emissions (SFOAEs) were recorded in the ear canal using a nonlinear procedure. The results showed that the nonlinear SFOAE to a brief tonal signal (4 kHz, 10 ms, 60 dB SPL) in a background noise (100–6000 Hz, 400 ms, 25 dB/Hz) increased in magnitude (with signal delay) over a similar time course to subjects' improvement psychophysically in detecting the same signal. Manipulation of the noise bandwidth revealed that the increases in SFOAE magnitude and the improvement in psychophysical detection both were caused primarily by off-frequency components of the noise: lowpass components for SFOAEs, and lowpass or highpass components for psychophysics. These findings are consistent with a change in the slope of the cochlear input-output function from highly compressive to less compressive over the duration of the background noise, a change that might be attributable to negative feedback from the efferent system. [Work supported by NIDCD.]

4pPP8. Parameters for the estimation of loudness from tone-burst otoacoustic emissions. Michael Epstein (Dept. of Speech-Lang. Path. and Aud., Ctr. for Commun. and DSP [ECE Dept.] Auditory Modeling and Processing Lab., Northeastern Univ., 1600 Massachusetts Ave., Boston, MA 02115, m.epstein@neu.edu) and Ikaro Silva (Northeastern Univ., Boston, MA 02115)

There is evidence that tone-burst otoacoustic emissions (TBOAEs) might be useful for estimating loudness, but appropriate analysis parameters for loudness estimation must first be examined. The purpose of the present work was to collect TBOAE measurements and loudness estimates across a wide range of levels in the same listeners. Both measures were recorded for 1- and 4-kHz stimuli and then analyzed using a wide range of parameters to determine which parameter set yielded lowest mean-square-error estimation of loudness with respect to a psychoacoustical, cross-modality-matching procedure and the inflected exponential (INEX) loudness model. The present results show strong agreement between 1-kHz loudness estimates derived from TBOAEs and loudness estimated using cross-modality matching, with TBOAE estimation accounting for a significant portion of the variance. Additionally, the results indicate that analysis parameters may vary within a reasonable range without compromising the results (i.e., the estimates exhibit some parametric robustness). The lack of adequate parametric optimization for TBOAEs at 4 kHz suggests that measurements at this frequency are strongly contaminated by the ear-canal resonances, meaning that deriving loudness estimates from TBOAEs at this frequency is significantly more challenging than at 1 kHz. [Work supported by Capita Foundation.]

4pPP9. The sensitivity of hearing by the resonance in *in vivo* human ear canal. Wei-De Cheng (Taiouan Interdisciplinary Otolaryngol. Lab., Chang Gung Univ.), Jen-Fang Yu (Chang Gung Univ.), Kuo-Wei Huang, Shang-Peng Chang (Chang Gung Univ.), and Chin-Kuo Chen (Chang Gung Memorial Hosp.)

This study is to discuss the sensitivity of human hearing sounds at different depths of ear canal, and to construct the measurement scales which can be the reference for the sound measurement and research in the future. Eighteen subjects aged from 20 to 30 years old with normal hearing and middle ears were studied. The pure tone audiometer and impedance audiometer were utilized to exam the frequency threshold of subjects. The real ear measurement was also utilized. The intensities of stimuli were 40, 50, 60, 70, and 80 dB SPL. The measured depths to the tympanic membrane were 0.5, 1.0, 1.5, and 2.0 cm, respectively. The gain of ear canal by different frequencies at 500, 1000, 2000, 4000 Hz was measured. Based on the results, there was moderate negative correlation between the resonance of ear canal and hearing. The larger the gain of ear canal was, the better the hearing of the subject would be. Consequently, the sensitivity of hearing for normal people would be affected by the resonance of ear canal. The resonance of each subject would be changed similarly at different measurement depths.

4pPP10. The effects of quinine on frequency selectivity, temporal resolution, and speech recognition in quiet and noise. Erica J. Williams and Sid P. Bacon (Dept. of Speech and Hearing Sci., Arizona St. Univ., PO Box 870102, Tempe, AZ 85287-0102, ejw@asu.edu)

Quinine causes a temporary disruption of outer hair cell (OHC) function, and thus can be used to examine the role of OHCs on auditory perception. In the present study, frequency selectivity, temporal resolution, and speech recognition were measured before, during, and after a quinine-induced hearing loss. Normal-hearing listeners ingested 5.76–11.43 mg/kg body weight of quinine, resulting in 5–15 dB of hearing loss. Frequency selectivity was estimated by comparing the level of a noise masker needed to mask a fixed-level, 2-kHz signal when the masker contained a spectral gap at 2 kHz and when it contained no gap. Similarly, temporal resolution was estimated by comparing the masker level needed to mask the 2-kHz signal when the noise masker contained a temporal gap and when it did not. Speech recognition thresholds were measured in quiet and in the presence of a masker (speech-shaped noise or time-reversed speech) fixed at 70-dB SPL. Signal level was varied adaptively to estimate 50% correct recognition. Quinine resulted in reduced frequency selectivity and reduced temporal resolution. Quinine also elevated speech recognition thresholds in quiet (by 7 dB on average), but the thresholds in the presence of the maskers were unaffected. [Work supported by NIDCD and AAA.]

4pPP11. Behavioral responses to harmonic complex tones with missing fundamental frequencies by chinchillas in the presence of low-pass masking noise. William P. Shofner (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, wshofner @indiana.edu)

The pitch associated with the missing fundamental (F0) is one of the principal psychological attributes of human pitch perception. Behavioral responses to missing F0 stimuli were measured in chinchillas using operant conditioning and stimulus generalization. Animals were trained to discriminate between harmonic complex tones having a 500-Hz F0 and a 125-Hz F0. When animals were tested with tone complexes having the same F0s, but where the F0s were missing, responses were similar to those obtained when the F0s were present, suggesting that missing F0 sounds were perceptually equivalent to F0 present sounds. In the presence of low-pass masking noise, responses to F0 present and missing F0 stimuli were similar, suggesting that the percept was not due to the reinsertion of the F0 through cochlear nonlinearities. When the F0s of test stimuli were systematically varied, gradients in behavioral responses were observed, suggesting the existence of a psychological dimension related to F0. When the F0 and spectrum were varied independently, responses were related to the F0 rather than to spectral differences among test stimuli. The results indicate that chinchillas possess a pitchlike perception of the missing F0 that is unlikely to arise from cochlear distortion products. [Work supported by NIDCD R01 DC005596.]

4pPP12. Temporal modulation transfer functions in listeners with real and simulated hearing loss. Joseph G. Desloge, Charlotte M. Reed, Louis D. Braida, Lorraine A. Delhorne, and Zachary D. Perez (Res. Lab. of Electron., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, jdesloge@mit.edu)

Temporal modulation transfer functions (TMTFs) were obtained in nine listeners with moderate to profound sensorineural hearing loss (age range of 21 to 69 years). The hearing loss of each impaired listener was simulated in a group of three age-matched normal-hearing listeners through a combination of spectrally-shaped masking noise and multi-band expansion. TMTFs in both groups of listeners were measured in a broadband noise carrier as a function of modulation rate in the range of 2 to 1024 Hz using a 3I, 2AFC procedure. The presentation level for the unmodulated broadband noise (whose duration was 500 ms) was set to be the maximum of either 70 dB SPL or the level such that noise was 10 dB above the lowest hearing threshold. The listeners with simulated hearing loss thus received signals at the same SPL and SL as their hearing-impaired counterparts. The shape of the TMTF curves (defined as the measured threshold of modulation in dB as a function of frequency of modulation) and the interpolated DC values of the function (using an exponential fitting procedure) were generally similar for listeners with real and simulated hearing loss. [Work supported by NIH-NIDCD R01 DC00117.]

4pPP13. Forward-masking functions in listeners with real and simulated hearing loss. Louis D. Braida, Joseph G. Desloge, Charlotte M. Reed, Lorraine A. Delhorne, and Zachary D. Perez (Res. Lab. of Electron., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, ldbraida@mit.edu)

Forward-masking functions were obtained in eight listeners with moderate to severe sensorineural hearing loss (aged 21–69 years). The hearing loss of each impaired listener was simulated in a group of three age-matched normal-hearing listeners through a combination of spectrally shaped masking noise and multiband expansion. Forward masking was measured in both groups of listeners for probe signals at frequencies of 500, 1000, 2000, and 4000 Hz using an on-frequency masker and two off-frequency maskers (0.55 and 1.15 times the signal frequency) under a 3I, 2AFC procedure. The probe signal was presented at 5 dB SL; signals and maskers were gated with 5-m on/off times with a steady-state duration of 0 ms for probe signals and 100 ms for maskers; and values of masker-offset time to signal-onset time were in the range of 0 to 100 ms. Findings will be described in terms of the slopes of the masking functions for each combination of probe and masker frequency and the ratio of the slopes obtained with on- relative to off-frequency maskers. Preliminary results suggest that the slope ratios observed for the hearing-impaired listeners were generally well-reproduced in normal-hearing listeners with simulated hearing loss. [Work supported by NIH-NIDCD R01 DC00117.]

4pPP14. A comparison of gap-detection thresholds through audition and touch in listeners with real and simulated hearing impairment. Charlotte M. Reed, Joseph G. Desloge, Zachary D. Perez, Lorraine A. Delhorne, and Louis D. Braida (Res. Lab. of Electron., Massachusetts Inst. of Technol., 77 Massachusetts Ave., Cambridge, MA 02139, cmreed@mit.edu)

Gap-detection thresholds were measured in listeners with real and simulated hearing loss under conditions of auditory or tactile presentation. The audiometric thresholds of each of ten listeners with sensorineural hearing loss (21–69 years of age) were simulated in groups of age-matched normal-hearing listeners through a combination of spectrally shaped masking noise and multiband expansion. The leading and trailing markers for the gap-detection task were 250- and 400-Hz sinusoids with a nominal duration of 100 ms. Gap-detection thresholds for a nominal baseline gap of 0 ms were measured for four different combinations of leading and trailing markers (250–250, 250–400, 400–250, and 400–400 Hz) using a 3I, 2AFC procedure. Auditory measurements were obtained for monaural presentation over headphones using a marker level set to be equal to the maximum of 70 dB SPL or 10 dB SL. Tactile measurements were obtained using sinusoids

presented to the left middle finger through an Alpha-M AV-6 vibrator at a level of 25 dB SL. Results are discussed in terms of (a) spectral-disparity effects of leading and trailing markers; (b) relation of thresholds for auditory versus tactile presentation; and (c) comparisons of results from listeners with real and simulated hearing loss. [Work supported by NIH-NIDCD R01 DC00117.]

4pPP15. Effects of several variables on temporal-order identification in young and elderly listeners. Daniel Fogerty, Diane Kewley-Port, and Larry Humes (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan, Bloomington, IN 47405, dfogerty@indiana.edu)

Four measures of temporal-order identification were completed by young (N=35, 18–31 years) and elderly (N=151, 60–88 years) listeners under various stimulus conditions. Experiments spaced over several months used forced-choice constant-stimuli methods to determine the smallest stimulus onset asynchrony (SOA) between brief (40 or 70 ms) vowels that enabled identification of the stimulus sequence. Vowels in four words (pit, pet, pot, put) served as stimuli. The four measures of temporal-order identification were: (1) monaural two-item sequences; (2) monaural four-item sequences; (3) dichotic two-item vowel-identification sequences; and (4) dichotic two-item ear-identification sequences. All listeners identified the vowels in isolation with better than 90% accuracy. Results indicated that elderly listeners performed significantly poorer on monaural and dichotic temporal-order identification tasks than young listeners, although a large overlap in group distributions was observed. For both groups, the two-item dichotic task was significantly harder than two-item monaural. Increasing the attentional demands of the monaural task by randomizing the stimulus ear did not explain this difference. Using shorter duration stimuli did not alter performance in the monaural task but did improve performance in the dichotic task. Significant learning occurred for elderly listeners but not enough to eliminate the age-group differences. [Work supported, in part, by NIA.]

4pPP16. Level-dependent changes in detection of a silent gap in fluctuating noise carriers. Amy R. Horwitz, Jayne B. Ahlstrom, and Judy R. Dubno (Dept. of Otolaryngol.-Head and Neck Surgery, Medical Univ. of South Carolina, 135 Rutledge Ave., MSC 550, Charleston, SC 29425, horwitar@musc.edu)

Changes in detection of a silent gap in a noise carrier with increasing carrier bandwidth and level may reveal effects of basilar-membrane nonlinearities on envelope fluctuations. Gap-detection thresholds are higher for narrowband than broadband noise carriers due to increased confusion between the imposed gap and inherent fluctuations of the narrowband noise. For narrowband more than broadband carriers, the "effective" magnitude of envelope fluctuations may be reduced by compressive effects of the active cochlear mechanism. To test these assumptions, detection of a gap was measured in 50-Hz-wide and 1000-Hz-wide noise carriers as a function of carrier level. Younger and older adults with normal and impaired hearing listened to carriers presented in quiet and in a low-fluctuation broadband masker. As carrier level increased, masker level also increased, maintaining a fixed difference between carrier and masker levels to minimize sensation-level effects on gap detection. For younger adults, gap detection measured in quiet improved as expected with increasing carrier level. Further, gap detection measured in the masker improved as carrier level increased for the 50-Hz carrier, but declined for the 1000-Hz carrier. Discussion will focus on effects of carrier level, bandwidth, subjects' age and thresholds, and hypothesized relations to basilar-membrane nonlinearities. [Work supported by NIH/NIDCD.]

4pPP17. Minimum audible angles in children who use bilateral cochlear implants. Cynthia M. Zettler, Shelly Godar, and Ruth Y. Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705)

Over 5000 children worldwide have received bilateral cochlear implants (CI) so that the auditory skills that rely on inputs to both ears, such as sound localization, may be improved. In this study, localization acuity was measured with minimum audible angles (MAA), the smallest discriminable distance between two locations in the frontal azimuth plane. It is unclear whether children who use bilateral CIs can attain MAAs comparable to their acoustically hearing peers, and if so, the extent to which exposure to bilateral hearing is necessary. Children having 3–36 months of bilateral experience participated. Stimuli were spondaic words presented at an overall level of 60 dB (± 4 dB rove). Every child with >5 yrs of auditory experience performed the task, whereas some of the children with <5 yrs of experience could not perform the task. In the former group, children with >2 yrs of bilateral experience were the best performers (MAA thresholds <15°), compared to children in the latter group who completed the task (MAA thresholds >30°). Results suggest that in children who receive bilateral CIs the two overall factors that impact performance are overall time in sound and amount of time with bilateral experience. [Work supported by NIH DC R01008365.]

4pPP18. The impacts of age and absolute threshold on binaural lateralization. Frederick J. Gallun, Anna Diedesch, and Erin Engelking (Natl. Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., Portland, OR, 97239, Frederick.Gallun@va.gov)

Listeners varying in audiometric thresholds (70-dB range) and in age (52-year range) lateralized complex tones relative to a diotic standard. Stimuli were composed of one, two, three, or six tones amplitude modulated at a rate of 75 Hz. Carrier frequencies were selected from a set of seven values (roughly log-spaced between 700 and 7000 Hz) and sensation levels were equated across listeners. Baseline performance was obtained for one- and two-component signals and the effects of diotic interference were assessed with two-, three-, and six-component signals. For all signals, the dichotic components contained interaural differences in time (whole-waveform delay), level, or both across a range of values. Listeners with higher audiometric thresholds performed more poorly in all conditions, despite the use of equivalent sensation levels. Diotic interferers reduced performance, but there were no additional impacts of age or absolute threshold. Statistical analyses indicated that the impacts of age were primarily due to the co-occurring increases in pure-tone thresholds, suggesting that: (1) absolute threshold is a better predictor of binaural ability than age and (2) reduced sensitivity to binaural differences is not addressed by presenting stimuli at equivalent sensation levels. [Work supported by VA RRD CDA-2 C4963W.]

4pPP19. Age-related differences in the time it takes to form auditory images of broadband noises. Mengyuan Wang (Dept. of Psych., Human Commun. Lab., Univ. of Toronto at Mississauga, 3359 Mississauga Rd. North, Toronto, ON L5L 1C6, Canada, mengyuan.wang.pku@gmail.com), Bruce Schneider (Univ. of Toronto at Mississauga, Toronto, ON L5L 1C6, Canada), Yanhong Wu, Xihong Wu, and Liang Li (Peking Univ., Beijing, 100871, China)

When correlated noises are presented over earphones to the two ears, listeners typically form a fused compact image of the noise. However, when the noises presented to the two ears are independent, listeners tend to hear two noises: one on the left and the other on the right. In Experiment 1 we determined the minimum duration required to form auditory images by asking listeners to distinguish between a 1-s presentation of independent noises to the left and right ears, and another 1-s presentation in which the noise was correlated for x ms before switching to two independent noises. In Experiment 2, one of the noises was correlated throughout the 1-s presentation; the other started off uncorrelated before switching to correlated. Younger adults performed better than older adults in both experiments. However, the performance of younger adults was better in Experiment 2 than in Experiment 1, whereas the reverse was true for older adults. The implications of these results for age-related changes in auditory scene analysis will be discussed. [This work was supported by the National Natural Science Foundation of China, the Natural Sciences and Engineering Research Council of Canada, and the Canadian Institutes of Health Research.]

4pPP20. Temporal processing as a function of age: Interaural time difference discrimination. John H. Grose, Sara K. Mamo, Emily Buss, and Joseph W. Hall (Dept. OHNS, Univ. N. Carolina at Chapel Hill, 170 Manning Dr., Chapel Hill, NC 27599-7070, jhg@med.unc.edu)

Older listeners often exhibit auditory temporal processing deficits. This study tested for temporal deficits in presenescent hearing. Three age groups participated: younger (18–28 yr); middle-aged (40–55 yr); older (63–75 yr). All had normal lower frequency hearing (thresholds ⩽ 20 dB HL, 250–2000 Hz). Exp. 1 measured ITD discrimination for tone frequencies from

250–1500 Hz (65 dB SPL). Stimuli were two 100-ms tone pulses (75-ms rise, 25-ms fall). The pulses were diotic in the standard stimulus; the signal had an ITD (random lead/lag) imposed on the trailing tone pulse. Older listeners generally had higher ITD thresholds than Younger listeners. Middle-aged and Younger listeners performed similarly at low frequencies but the middle-aged tended to hit ceiling (ʒ radians) at lower frequencies. Exp. 2 measured discrimination of So and Sτ signals as a function of frequency. Stimuli were tonal carriers modulated at 5 Hz (4 periods). Standard AM tones were diotic; signal tones alternated between So and Sʒ during successive modulator periods. Middle-aged listeners often performed slightly poorer than younger listeners, but better than older listeners. These results suggest that temporal processing deficits are evident in the pre-senescent auditory system. [Work supported by NIDCD 5-R01-DC01507.]

4pPP21. Quality judgments for music signals by normal-hearing and hearing-impaired listeners. Kathryn H. Arehart (Dept. Speech Lang. and Hearing Sci., Univ. of Colorado, Boulder, CO 80309), James M. Kates (GN ReSound and Univ. of Colorado, Boulder, CO 80309), and Melinda C. Anderson (Univ. of Colorado, Boulder, CO 80309)

Noise, distortion, nonlinear signal-processing algorithms, and linear filtering can all affect the sound quality of a hearing aid or other audio device. Most hearing-aid research concentrates on speech, but music reproduction can also be an important factor in user satisfaction. In this presentation, quality judgments are made for several different music signals by normal-hearing and hearing-impaired listeners. The music signals include orchestral classical music, jazz instrumental, and vocal. The signal processing uses a simulated hearing aid. The noise and nonlinear signal degradations include additive noise, multitalker babble, peak-clipping distortion, quantization noise, multichannel compression, spectral subtraction. Linear filtering conditions include bandwidth limitation, spectral resonance peaks, and spectral tilt. Conditions combining noise and nonlinear processing with linear filtering are also included. Subject ratings of the degraded music will be presented, along with comparisons of the ratings of music with that of speech for the same set of processing conditions. In addition, the subject ratings will be compared predictions using quality metrics based on auditory models.

4pPP22. A speech quality metric based on a cochlear model. James M. Kates (GN ReSound and Univ. of Colorado, Boulder, CO) and Kathryn H. Arehart (Univ. of Colorado, Boulder, CO)

Noise, distortion, nonlinear signal-processing algorithms, and linear filtering can all affect the sound quality of a hearing aid. The sound quality, in turn, can have a strong impact on the success of the device. The general approach in designing a quality metric is to compare the degraded signal with a clean (unprocessed) reference signal; the comparison generally involves comparing one or more features extracted from the signals. In this presentation, features are extracted from a computationally efficient model of the auditory periphery. The model includes the middle ear, an auditory filter bank, dynamic-range compression, two-tone suppression, and loudness scaling. The first step is a metric for noise, distortion, and nonlinear signal processing. The noise and nonlinear metric focuses on differences in the short-time signal behavior. The second step is a metric that focuses on the changes in the long-term average spectrum caused by linear filtering. The third and final step is to merge the noise and nonlinear metric with the linear filtering metric to produce a composite sound quality metric that can be applied to an arbitrary signal-processing system. The metrics give correlation coefficients better than 0.94 in comparison with quality ratings made by normal-hearing and hearing-impaired listeners.

4pPP23. Preference and performance for hearing-aid gain-compression prescriptions, based on a preliminary model of impaired sensitivity to tone intensity. Julius L. Goldstein (Hearing Emulations LLC, at Ariel Premium, 8825 Page Ave., St. Louis, MO 63114, julius@hearem.com), Michael Valente (Washington Univ., St. Louis, MO 63110), Metin Oz, and Peter Gilchrist (formerly at Hearing Emulations LLC)

A cochlearlike preliminary model was synthesized by joining a model of loss of cochlear "tip" gain, representing mild hearing loss, with a model for greater loss that translates the residual range for tone intensities into nonlinear "tail" expansion. Normal tip gain was modeled as 40 dB with square-root compression. Tail expansion was modeled with upper bounds for residual hearing at normal LDL of 100 dB HL, or at levels increasing by half decibel per decibel hearing loss above 40 dB; the latter provides less compression. A digital hearing aid comprising six octave-bandwidth (0.25–8 kHz) linear amplifiers with AGC was simulated to test alternative prescriptions with 32 experienced users of hearing aids. Preference for prescriptions with high or low compression and 4 or 8 kHz bandwidth was tested with clean speech repeated at input levels of 40, 60, and 80 dB SPL. The alternative best perceived as "mild," "comfortable," and "loud" was chosen; if uncertain, "either" was chosen. Performance was measured with intelligibility scores for low-probability speech sentences in noise (SPIN) at 8 dB signal to noise ratio and 70 dB SPL input. Independent of preference, high-compression prescriptions provided inferior performance, resulting from excessive increase in compression with hearing loss. [Work supported by NIDCD.]

4pPP24. A cochlearlike model of equal loudness levels for impaired and normal hearing: A new basis for hearing-aid gain-compression prescriptions. Julius L Goldstein (Hearing Emulations LLC, at Ariel Premium, 8825 Page Ave., St. Louis, MO 63114, julius@hearem.com)

Mean psychophysical data on most comfortable and uncomfortable tone levels as a function of hearing loss [Pascoe, 13th Danavox Symp. 153–183 (1988); D. M. Schwartz *et al.*, The Hearing J. **41**, 13–17 (1988)] is interpolated for other loudness levels by representing impaired auditory response as a modification of the idealized normal cochlear mechanical response to tones: linear at low and high sound levels joined by power-law compression of the low level "tip" gain. Denoting the low- and high-level nonlinear transitions as stimulus thresholds for compression and decompression, hearing loss elevates the compression threshold without changing the decompression threshold, until the compressive region is eliminated. Further hearing loss converts the compression threshold into a threshold for power-law "tail" expansion. Mean psychophysical data, represented by multiple regression lines, are fit exactly by an analytical solution for the normal model parameters. Estimates for thresholds of approximately 15 and 95 phons and tip gains of 40–52 dB are physiologically plausible for the normal cochlea. Normal equal loudness levels updated by Suzuki and Takeshima [Y. Suzuki and H. Takeshima, J. Acoust. Soc. Am. **116**, 918–933 (2004)] allow reliable extension of the model to low frequencies with estimates of auditory high-pass linear acoustic processing and of normal low-frequency tip gains. [Work supported by NIDCD.]

4pPP25. A weighting-function-based approach to subjectively modify the frequency response of a hearing aid. Andrew T. Sabin, Nicole Marrone, and Sumitrajit Dhar (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60201, a-sabin @northwestern.edu)

While approaches that modify the frequency response of a hearing aid based on a listener's subjective preference have demonstrated some success, they also have several limitations. Namely, these approaches (1) consider a small set of potential frequency-gain curves (FGCs) and (2) they converge on the final FGC, which makes the outcome dependent upon the starting point. Here we present a new approach that addresses these problems by (1) considering a much larger initial set of FGCs, and (2) using a method that is not convergence based. First, the listener rates the clarity of short samples of speech filtered by a set of maximally different probe FGCs. A weighting function is then computed, where the weight given to each ~1/3 octave frequency band is proportional to the normalized slope of the regression line between the listener's rating and the within-band gain of the probe FGC. Next, the listener rates the clarity of speech samples filtered by the weighting function, which is multiplied by one of several, randomly ordered, scaling coefficients. The final FGC is the scaling coefficient that receives the maximum rating, multiplied by the weighting function, added to a hearing-loss-specific correction factor. Results will be compared to current fitting strategies.

4pPP26. Effects of aging and irrelevant-dimension fluctuation on frequency doscrimination: Interference at late information-processing stages. Praveen Jajoria (Medical Sch., Univ. Texas Medical Branch, Galveston, TX 77555–0570) and Blas Espinoza-Varas (Dept. Commun. Sci. & Disord., Univ. Oklahoma Health Sci. Ctr., 1200 N. Stonewall, Oklahoma City, OK 73104)

Deficits in the ability to extricate relevant from irrelevant information at the perceptual level (i.e., selective attention) could help diagnose age-related cognitive decline at an early stage. Much experimental evidence supports this generalization, but there is debate as to where and how the irrelevant information interferes with the relevant one. To address this question, effects of irrelevant duration or level differences on pure-tone frequency discrimination thresholds (FDTs) were studied in healthy, normal-hearing, young and older adults. Target tones were presented either in isolation or followed by a distracter tone after a 90- or 350-ms silent interval; both tones were 1000 Hz, 80 ms, and 70 dB SPL. Irrelevant differences occurred simultaneously with relevant frequency differences or sequentially, in the distracter. With both presentation formats and silent intervals, FDT elevations (i.e., interference) were much larger in older than in young adults if the irrelevant differences were in level, but if they were in duration, FDT elevations were small in both groups. Since it occurs both simultaneously and retroactively (i.e., sequential), and depends on the similarity of relevant and irrelevant dimensions, the interference seems to take place at relatively late rather than early information-processing stages.

4pPP27. Dual-task costs in speech recognition by older and younger listeners. Karen S. Helfer, Jamie Chevalier, and Richard L. Freyman (Dept. of Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01003)

This study examined performance of younger and older adults when they carried out two auditory tasks simultaneously versus when performing each task in isolation. One task was to repeat a target sentence spoken by a male talker, presented simultaneously with two masking sentences spoken by two other male talkers. The second task was to judge whether neither, one, or both voices in the two-talker masker complex was presented time-reversed. Data were collected in both spatially-separated and colocated conditions. Results showed substantial individual variability within both subject groups as well as a strong effect of trial type order. Performance by most listeners declined to a greater extent for the natural/time-reversed task than for the speech recognition task during divided attention, even though subjects were instructed to give the two tasks equal weight. Importantly, requiring listeners to divide attention decreased or eliminated the spatial separation advantage for listening to speech in a competing speech situation. The poster will discuss differences noted between the two groups in overall performance as well as in the costs of dividing attention. [Work supported by NIH DC01625.]

4pPP28. Perceptually important acoustic features of environmental sounds. Laurie M. Heller, Benjamin Skerritt, and Emily Ammerman (Dept. of Cognit. and Linguistic Sci., Brown Univ., Box 1978, Providence, RI 02912, Laurie_Heller@brown.edu)

Listeners were asked to indicate the sources of dozens of sound events presented over headphones. The sounds were environmental sounds made by solid objects, liquids, and/or air undergoing a variety of actions taken from the website www.auditorylab.org. Each listener gave a series of responses about possible sources for each sound so that a similarity matrix could be constructed. The results of data reduction point to a manageable set of perceptual classes of sound events that can be described with a small number of distinctive spectro-temporal acoustic features. These features will be related to the attributes of the events that generated each class of sounds. Results for normal-hearing listeners will be analyzed in detail and preliminary results from hearing-impaired listeners will be previewed. [Work supported by NSF 0446955 and RI STAC.]

4pPP29. The incongruency advantage in elderly versus young normal-hearing listeners. Brian Gygi (Speech and Hearing Res., Veterans Affairs Northern California Health Care System, Martinez, CA) and Valeriy Shafiro (Rush Univ. Medical Ctr., Chicago, IL)

Previous research [Gygi & Shafiro (2007); Leech *et al.* (2007)] reported that environmental sounds heard in contextually incongruous naturalistic auditory scenes are better identified than sounds contextually congruous with the scene (e.g., rooster crowing in a farm ambience versus rooster in an office ambience). This incongruency advantage (IA) averages about four-to-five percentage points in p(c) and has been observed in both well-trained and naïve young normal hearing listeners [Gygi and Shafiro (2008)] and in children [Leech *et al.* (2007)]. One aspect of the IA is that it appears to be level-dependent, in that the effect is not present for lower sound/scene ratios (So/Sc). For experienced young listeners, the IA appears at a So/Sc of -15 dB. However, for naive young listeners, it is not manifest until -7.5 dB. A group of elderly normal-hearing listeners were tested and exhibited an IA only at So/Sc of -9 dB and above. The results show that the incongruency advantage are complex effects resulting from a combination of peripheral and central factors. [Research supported by the Merit Review Training Grant from the Dept. of Veterans Affairs Research Service, VA File # 06-12-00446, the National Institute for Aging, and the Rush Univ. Medical Center.]

THURSDAY AFTERNOON, 21 MAY 2009 — FORUM SUITE, 1:25 TO 2:20 P.M.

Session 4pSAa

Structural Acoustics and Vibration: Vibro-Acoustic Diagnosis and Prognosis of Complex Structures II

Wen Li, Chair

Dept. of Mechanical Engineering, Wayne State Univ., Detroit, MI 48202

Invited Paper

1:25

4pSAa1. Synchronization of mechanical phase oscillators. David Mertens, Richard Weaver (Dept of Phys., Univ. of Illinois, 1110 W. Green St., Urbana, IL), and John Koliniski (Div. of Eng. and Appl. Sci., Harvard Univ., Cambridge, MA)

The Kuramoto model of a large number of weakly interacting phase oscillators exhibits a phase transition to synchronization. The literature has seen analytic theory and numerical simulations for both the Kuramoto model and sundry generalizations. Most real-world systems (e.g., flashing fireflies) imperfectly match the model, and their synchronization behaviors can therefore be taken to be in merely qualitative support of the theory. Here a mechanical system with well-understood microphysics is presented. A number N (of order 50) of eccentrically weighted DC motors (cell phone vibrators) is mounted on a plate. Each motor radiates into the plate and responds to its motion. If the plate dynamics is dominated by a single normal vibration mode the system is well described by finite-N Kuramoto equations. Transitions to synchronization are observed in accord with theoretical expectations based on the quality factor of the plate, the number of motors, the ratio of the motor mass to the plate mass, and the natural speed of the motors. Calculations and results on a rich generalized-Kuramoto model are also presented, in which the plate dynamics entails many normal modes of vibration, and for which the governing equations resemble those of a laser.

Contributed Papers

1:50

4pSAa2. Analysis and prediction of underwater sound radiation from a cylindrical pile driven by an impact hammer. Mardi C. Hastings (Appl. Res. Lab., Penn State Univ., P. O. Box 30, State College, PA 16804, mch26@psu.edu)

A vibroacoustic model for sound radiation from a submerged pile was developed to predict sound pressure levels generated during offshore construction activities. It was implemented in MATLAB and verified with measured data reported in the literature. The modal response of a cylindrical pile to hammer impact is first determined. Contributions from all dominant modes are then summed at various ranges to produce a two-dimensional spatial mapping of sound pressure level. The analysis was used to investigate the effects of pile material, size, and geometry on radiated sound pressure levels. Results indicate that smaller diameter piles of the same material do not necessarily produce lower sound pressure levels, but rather shift the peak sound pressure levels to higher frequencies. Because these basic parameters affect bandwidth as well as sound levels, there is no simple way to classify source level and frequency. Peak sound pressure levels generated underwater by driving steel piles with impact hammers into the sea bottom can easily exceed 200 dB re 1 μPa at a range of 10 m, so accurate prediction of the sound field is needed to assess potential risks to the environment.

2:05

4pSAa3. On vibroacoustic modal analysis of arbitrarily shaped vibrating structures. Huancai Lu (SeeSound 360, 3649 Glenwood Ave., Windsor, ON N9E 2Y6, Canada, huancai.lu@gmail.com) and Sean Wu (Wayne State Univ., Detroit, MI 48202)

Vibroacoustic modal analysis based on Helmholtz least squares method (HELS) is presented in this paper to explore the inherent dynamic characteristics and correlation of structural vibration and sound radiation of arbitrarily shaped vibrating structures. A series of mutually orthogonal vibroacoustic components is established by decomposing the normal velocity and normal acoustic intensity, which are reconstructed on source surface using HELS, in an orthogonal space through singular value decomposition (SVD). By further analyzing the vibration efficiency and radiation efficiency, the contribution of each of individual modal components to the resultant structural vibration and sound radiation can then be quantified. Therefore one is allowed to identify the vibroacoustic modal components that are most responsible for resultant structural vibration and sound radiation. The test object is a clamped and baffled thin rectangular steel plate, which was excited by a shaker and tested in a semianechoic chamber. The reconstructed normal surface velocities of the plate are validated against the results scanned by laser equipment. The vibration efficiency and radiation efficiency of the vibroacoustic modal components are examined, and the correlation between structural vibration and sound radiation of components at several frequencies is analyzed.

THURSDAY AFTERNOON, 21 MAY 2009 — FORUM SUITE, 2:25 TO 3:55 P.M.

Session 4pSAb

Structural Acoustics and Vibration and Engineering Acoustics: Concepts of New Vibration Sensors

Daniel W. Warren, Chair
Knowles Electronics Inc., 1151 Maplewood Dr., Itasca, IL 60143

Invited Papers

2:25

4pSAb1. Role of slowness mapping in determining the directions of acoustic and seismic signals. Qamar A. Shams, George R. Weistroffer, John W. Stoughton (NASA Langley Res. Ctr., M.S. 238, Hampton, VA 23681), and Allan J. Zuckerwar (Analytical Services Mater., Hampton, VA 23666)

Slowness mapping is a method to estimate the angle of arrival of plane waves propagating across a sensor array. A review of time-delay estimation and its application to slowness vector estimation, the forward model, the inverse model, azimuth estimation, and elevation estimation will be presented. A method for performance grading with "out-of-bounds" conditions is described, and in the special case of subsurface acoustic sensors, a method for discriminating against seismic signals. The method has been applied to locate the direction of Space Shuttle and other rocket launches, infrasonic emissions from clear air turbulence, and incidental sources found in the environment.

2:50

4pSAb2. Experimental validation of alternate integral-formulation method for predicting acoustic radiation based on particle velocity measurements. Zhi Ni and Sean Wu (Wayne State Univ., 5050 Anthony Wayne Dr. Detroit, MI 48202)

This paper presents experimental validation of an alternate integral-formulation method (AIM) for predicting acoustic radiation from an arbitrary structure based on the particle velocities measured by a laser Doppler anemometer (LDA) and double hotwire sensor over a hypothetical surface enclosing a target source. Both the normal and tangential components of the particle velocity on this hypothetical

surface are measured and taken as input data to AIM codes to predict acoustic pressures in the exterior region. The results thus obtained are compared with those measured by microphones at the same locations. Measurement limitations are discussed and an error analysis of LDA measurement is presented.

3:15

4pSAb3. Visualization of vibrations measured with a multi-channel optical vibration sensor. Jun Hasegawa and Kenji Kobayashi (Dept. of Electron. and Comp. Syst., Takushoku Univ., 815-1, Tatemachi, Hachioji-shi, Tokyo 193-0985, Japan, jhase@es.takushoku-u.ac.jp)

Displacement-type sensor units made of optical fibers were developed to realize multi-points measurement of vibrations with high spatial resolution. Each sensor unit has the displacement resolution of 10 nm within the dynamic range of more than 90 dB, the frequency bandwidth of up to 80 kHz. Up to 64 sensor units can be arrayed as a multi-channel sensor head, with the minimum gap between sensor units of 4 mm. It means that the spatial resolution for the multi-points measurement is 4 mm. The calibrating method with the measuring object was developed to realize accurate measurements. During the calibration phase, the object is kept stand-still, and only the sensor head moves by the linear actuator. Thus, the calibration data can be obtained just before the measurement. Several arrangements of the sensor system is available depending on the object to be measured, such as line vibration, surface vibration, and the 3D movement of small object. The developed system has been used for the measurement of the actuators for vibratory microinjection and the measurement of artificial heart valves. Those results showed the advantage of the system. [This study has been supported by JSPS under the Grants-in-Aid for Scientific Research, Nos. 14550427 and 16560369.]

Contributed Paper

3:40

4pSAb4. Acoustic sensor structural configuration study. Bill B.B. Zhang and W. Steve Shepard, Jr. (Dept. of Mech. Eng., The Univ. of Alabama, Box 870276, Tuscaloosa, AL 35487, sshepard@eng.ua.edu)

This work examines the configuration requirements for a sensor being developed to provide information about the location of an acoustic source. The continuous sensor's vibrational response, caused by a traveling acoustic wave, is used in an inverse method to reconstruct the forces in that wave. A finite-length beam supported on an elastic foundation is the candidate structure. The impact of changing various structural parameters on the beam's vibration response is examined analytically using finite element methods. A goal in configuring the beam sensor is to obtain a response with enough dynamic information to accommodate a successful force reconstruction. Some of the configuration parameters of interest include the beam material, geometry, thickness, and the elastic foundation properties. The excitations used in the study consist of transient waves with differing numbers of sinusoidal half-cycles. Because background noise has the potential to impact the force reconstruction, various amounts of white noise are added to the simulated response values prior to performing the traveling wave reconstruction. Tikhonov regularization and the L-curve method are used. Results obtained from the simulations show that this sensor model is effective in identifying moving wave loads. [Work supported by NSF Sensor Innovation and Systems.]

THURSDAY AFTERNOON, 21 MAY 2009 — GRAND BALLROOM I, 1:00 TO 4:00 P.M.

Session 4pSC

Speech Communication: Second Language and Cross-Linguistic Speech (Poster Session)

Megha Sundara, Chair

Dept. of Linguisitcs, Univ. of California at Los Angeles, Los Angeles, CA 90095-1543

Contributed Papers

All posters will be on display from 1:00 p.m. to 4:00 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 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

4pSC1. An acoustical comparison of English tense and lax vowels. Byunggon Yang (Dept. of English Education., Pusan Natl. Univ., 30 Changjundong Keumjunggu, Pusan 609-735, South Korea, bgyang@pusan.ac.kr)

Several studies on the pronunciation of English vowels point out that Korean learners have difficulty producing English tense and lax vowel pairs. The acoustic comparisons of those studies are mostly based on the formant measurement at one time point of a given vowel section. However, the English lax vowels usually show dynamic changes across their syllable peaks and subjects' English levels account for various conflicting results. The purposes of this paper are to compare the temporal duration and dynamic formant tracks of English tense and lax vowel pairs produced by five Korean and five American males. Results showed that both the Korean and American males produced the vowels with comparable durations. The duration of the front tense-lax vowel pair was longer than that of the back vowel pair. From the formant track comparisons, the American males produced the tense and lax pairs much more distinctly than the Korean males. The results suggest that the Korean males should pay attention to the F1 and F2 movements, i.e., the jaw and tongue movements, in order to match those of the American males. Further studies are recommended on the auditorily acceptable ranges of F2 variation for the lax vowels.

4pSC2. The production of new versus similar vowels by Korean-English bilingual children. Soyoung Lee (P.O. Box 413, Enderis Hall 855, Univ. of Wisconsin-Milwaukee, 2400 E Hartford Ave., Milwaukee, WI 53201) and Gregory Iverson (Univ. of Maryland, College Park, MD 20742-5121)

The purpose of this study was to compare the vowel productions of Korean-English bilingual children with those of monolingual English and Korean-speaking children. Flege (1987) hypothesized that equivalent clas-

sification prevents adult L2 learners from establishing separate phonetic categories for similar phones (which occur in both languages) but not for new L2 phones. This hypothesis, however, has not been fully tested in bilingual children. The present study examined twenty 5-year-old Korean-English bilingual children who had been exposed to both languages for at least 2 years along with their age-equivalent monolingual counterparts, for a total of 60 children. Nine English vowels and six Korean vowels were elicited using picture cards. First and second formant frequencies for similar (/i, ɛ, u, o, a/) and new (/ɪ, ɔ, ɨ, æ, i/) vowels were measured. Vowels which were categorized as new were not appreciably different from those of monolinguals, suggesting that bilingual children produce new vowels authentically, consistent with the Flege hypothesis. However, the findings were mixed with respect to similar vowels. Bilingual children produced English vowels similarly to monolingual English-speakers, but their Korean vowels differed from those of monolingual Korean children, implying influence of the dominant (English) over the less-dominant language (Korean).

4pSC3. Acoustic features and inteilligibility of American-English vowels for English, Chinese, and Korean talkers. Su-Hyun Jin, Chang Liu, and Sangeeta Kamdar (Dept. of CSD, Univ. of Texas Austin, 1 University Station, Austin, TX 78712)

Sixteen American-English vowels including 12 monothongs and 4 diphthongs were recorded in a phonetic context of /hVd/ from young English (E), Chinese (C), and Korean (K) talkers. The Chinese and Korean talkers were bilingual and had stayed in United States up to 6 years. Two sets of experiments will be discussed: acoustic analysis and intelligibility of English vowels produced by the three groups of talkers. Results of acoustic analysis showed that there was no significant difference in F1×F2 vowel space among the three groups of talkers. In addition, the three groups of talkers showed great similarity in F2/F1 ratio across the 12 monothongs. Vowel durations had significantly greater variability across vowel categories for the Chinese and Korean talkers than for the English talkers, indicating that, besides producing spectral differences among vowels, Chinese and Korean talkers also attempted to generate durational difference among vowels to make each vowel distinguishable from others. More acoustic features such as spectral tilt and formant transition in the diphthongs and the effects of acoustic features on vowel perception by native English listeners will be discussed. Furthermore, the relationship between the vowel intelligibility and the second language experience of non-native talkers will be examined.

4pSC4. How regional dialect effects second language learning. Mu-Ling Teng (No. 1001, Univ. Rd., Hsinchu, Dept. of Foreign Lang. and Lits, Natl. Chiao Tung Univ., Taiwan 300, ROC), Yi-Chu Ke, Chu-ting Chen, Bo-hong Lu, Lai-Iok Ip, Ho-hsien Pan (Natl. Chiao Tung Univ., Taiwan), Shih-wen Chen (Natl. Tsing Hua Univ., Taiwan), and Hsiu-Min Yu (Chung Hua Univ., Taiwan)

This paper discussed dialect effects on L2 learning. Data from advanced English-learning students were separated into groups according to geographic regions: Northern, Middle, and Southern parts of Taiwan. Subjects were asked to read out loud eight English words, namely, "heed," "hid," "head," "had," "hod," "hawed," "hood," and "who'd." Acoustic information, such as F1 and F2, were measured; also, since Taiwanese English learners tend to use temporal cues to distinguish tense and lax vowels, duration of each word were also measured. Comparing the vowels read by the subjects with native speakers, we found that the Northern group had very similar patterns with native speakers in both spectral and temporal cues. As for the Middle group, the vowel shape was going upper right in the vowel chart and the duration pattern was quite distinctive from native speakers. Furthermore, all the subjects from the Southern part of Taiwan speak both Mandarin and Taiwan Min and their vowel spaces were the most deviated from native speakers, from which we assumed that PAM model (Best, 1993, 1995) could give a reasonable explanation: they have two vowel systems that could be referenced from, thus they might easily get confused. However, several problems still need further examination.

4pSC5. Production of prosodic cues by Beijing Mandarin speakers in second language (L2) English. Karen A. Barto-Sisamout (SLAT Office, P.O. Box 210014, Univ. of Arizona, Tucson, AZ 85721, kabarto@email.arizona.edu)

Does the prosody of speakers' first language (L1) influence their prosody in their second language (L2)? The current work investigates this topic for a tone language, Beijing Mandarin (BMan), as L1, and an intonation/stress language, English, as L2. English uses F0 contours as part of the intonation system, to signal syllabic prominence in a word, word prominence in a phrase, and the difference between questions and statements. In English there is a pitch peak delay, where the F0 peak occurs after the stressed syllable in two-syllable stress-initial words. Conversely, BMan uses F0 contours lexically, as a lexical tone language, and the F0 peak is on the stressed syllable. The production of these F0-related prosodic cues is investigated with BMan and Native English production of English narrow and broad focus statements and questions, with measurements of F0 contours on accented and unaccented words, pitch peak delay on accented words, and F0 levels in statements and questions. The goal is to learn if BMan speakers lexically specify tone of certain word types in English like in their L1, or realize F0 contours intonationally, like English speakers, or, finally, if they use an intermediate system different from L1 and L2.

4pSC6. Limited effects of early language learning on second language speech production. Tetsuo Harada (School of Education, Waseda Univ., 1-6-1 Nishi Waseda, Shinjuku, Tokyo 169-8050, Japan, tharada@waseda.jp)

Knightly *et al.* (2003) report long-term production benefits of childhood second language (L2) exposure in a naturalistic setting. However, it is unknown whether or not their finding will apply to classroom L2 learning. This study compares the production of voice onset time (VOT) in English and Japanese and closure duration of singletons and geminates in Japanese by English-speaking university students ($n=15$) who graduated from a Japanese immersion program (early learners) and adult learners of Japanese ($n=15$) with no exposure to Japanese in childhood (late learners). The informants were asked to repeat several target words including initial /p, t, k/ for VOT, and medial /p, t, k/ and /pp, tt, kk/ for singletons and geminates in a sentence frame. Both VOT of the initial stops and closure duration of the medial stops were measured. The results show that the production of VOT and closure duration by the early learners was not significantly different from that of the late learners ($p=0.71$ and $p=0.13$, respectively). The findings may hypothesize that there are only limited effects of classroom L2 exposure in childhood on L2 speech production in adulthood. [Work supported by Grant-in-Aid for Scientific Research (C) 20520527.]

4pSC7. The effect of oral proficiency on production of rhythm in spontaneous second language (L2) Japanese speech. Irina Shport (Dept. of Linguist., Univ. of Oregon, Eugene, OR 97403, ishport@uoregon.edu)

This study addresses the question of whether oral proficiency in Japanese second language (L2) speech has a unique correlation with acoustic characteristics of rhythm production that is independent from segments. Among four rhythm measures used (V%, VarcoV, VarcoC, VI-M), only two measures were different for the spontaneous L2 Japanese speech of beginning and intermediate learners. The interlanguage rhythm of less proficient speakers of Japanese was characterized by lower variability in duration of vocalic stretches (VarcoV) and higher variability in duration of consonantal stretches (VarcoC), $p < 0.05$. For both VarcoV and VarcoC values, the distribution of the individual speakers' rhythm scores was much tighter and on target for the intermediate students than for the beginning students. Furthermore, VarcoV values were significantly correlated with number of utterance-final vowels, and VarcoC values were correlated with number of obstruent clusters. In sum, the findings suggest that rhythmic differences in spontaneous L2 speech have an epiphenomenal nature stemming from the segmental structure of Japanese: the acquisition of mora-timed rhythm by learners of Japanese seems to be contingent upon the target-like production of segments which varies with proficiency level of learners.

4pSC8. Cross-linguistic differences in prosodic organization: Evidence from a repetition task. Emily Nava and Louis Goldstein (Dept. of Linguist., Univ. of Southern California, GFS 301, Los Angeles, CA 90089)

The prosody of languages such as English and Spanish has been characterized as exhibiting different rhythmic organizations. English has been hypothesized to organize syllables into feet, with one stressed syllable per foot. Spanish is among the languages whose prosody has been hypothesized to not include the foot, despite the existence of lexical stress in Spanish. We tested the hypothesis that these potential differences in organization would lead to systematically different responses when speakers were asked to entrain a sequence of syllables with a metronome at an increasing rate, and that L1Spanish/L2English speakers would continue to exhibit the Spanish pattern in their English. Speakers of all three types were recorded producing a single repeated syllable, or a sequence of two alternating syllables, in time with a metronome, whose rate increased monotonically after a stabilization period. At slower speech rates, English speakers produced each word as a separate foot with a corresponding pitch accent, while at increased speech rates, two words were grouped together into a single iambic foot with one pitch accent. Spanish speakers and L1Spanish/L2English speakers are expected to shorten both vowels, employing a symmetric strategy (as opposed to the asymmetric strategy of English speakers) to keep pace with the metronome. [Work supported by NIH, NSF.]

4pSC9. Effect of word length on vowel production by Mandarin and American speakers: Comparison of [i] and [ɪ]. Chung-Lin Yang (Dept. of Linguist., Indiana Univ., Bloomington, IN 47405, cy1@indiana.edu)

Yang (2008) found that Mandarin speakers' productions of [eɪ] and [ɛ] (as in *gate-get*) showed much less difference than Americans made. In addition, they made a smaller distinction in disyllables than in monosyllables. The current study compares the production of [i] and [ɪ] (*beat-bit*) in the same conditions. Target-vowels were embedded in Stop-V-Stop context in carrier sentences. Vowel durations were measured after stop release until final closure. Formant measurements were made at two time-points for each token. Preliminary analysis extends previous results of reduced contrast between tense and lax vowels by the Mandarin speakers to vowels [i]-[ɪ] and also shows greater reduction of the contrast in disyllabic words relative to monosyllables. The Mandarin speakers' formants for [i]-[ɪ] were almost identical in the disyllabic words, and their duration ratios were not significantly different, whereas American speakers' measurements were very different between two vowels and the same in disyllables as monosyllables. The current study provides further evidence for an effect of word length on English vowel contrasts by Mandarin speakers.

4pSC10. Short-term cross-linguistic interactions in bilinguals' vowel production. Matthew T. Carlson, Howard Nusbaum, and Shannon L. Heald (Dept. of Psychol., Univ. of Chicago, 5848 S. Univ. Ave., Chicago, IL 60637, carlsonmt@uchicago.edu)

Evidence suggests that the languages of a bilingual interact, and research has examined how first language (L1) phonetic categories influence second language (L2) acquisition. Although there may be substantial cross-linguistic interaction in the early stages of L2 learning, less is known about acoustic-phonetic variability when the bilingual's systems are more stable. This study examines L1 and L2 phonetic interaction, testing how the production of vowels in one language affects vowel production in the other. L1 and L2 vowel productions are compared for late English-Spanish bilinguals in an English-dominant environment. Five Spanish vowels and their English equivalents are examined. Subjects completed two experimental sessions, focusing respectively on L1 and L2, on separate days. Each session contained three blocks in which subjects produced vowels in isolation: two blocks in the target language separated by a block in the non-target language. Word cues were given to ensure the proper language for each block. Productions from first and third blocks of each session (i.e., in the same language) are compared to determine if intervening use of the other language affects subsequent vowel production. Implications are discussed for L1 and L2 phonetic representation, the status of a bilingual's languages, and second language acquisition.

4pSC11. Individual differences in the perception of vowels in a second language. A. Lengeris (Dept. of Speech Hearing and Phonetic Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, UK, a.leggeris@ucl.ac.uk)

Individuals may differ in their ability to learn the sounds of a second language (L2), but the origin of this variability remains uncertain. The present study examined whether individual differences in L2 vowel processing are related to individual differences in L1 vowel and/or nonspeech processing. Greek learners of English were given a large battery of perceptual tests examining their (1) identification of natural Greek vowels in noise, (2) identification and discrimination of synthetic Greek vowels in quiet, (3) identification of natural English vowels in quiet and in noise, (4) identification and discrimination of synthetic English vowels in quiet, and (5) discrimination of a nonspeech (F2 only) continuum in quiet. Preliminary results show a large degree of variability between individuals not only in L2 but also in L1 and nonspeech tasks. However, despite this variability, the participants were consistent in their performance across speech and nonspeech tasks, a finding that supports an auditory acuity rather than a speech-specific explanation for the individual differences in performance found in the data.

4pSC12. Perception of second-language stress and vowel quantity by English learners of Czech. Václav J. Podlipský (Dept. of English and American Studies, Palacký Univ., Krizkovskeho 10, 77180 Czech Rep., vaclavjonaspodlipsky@centrum.cz)

L2 acquisition may involve reattuning the perceptual system so that a cue becomes utilized for a new linguistic purpose. For instance, if vowel duration cues stress in L1 (like in English), whereas in L2 it marks phonological vowel quantity (like in Czech) it is of interest how the perception of stress and of vowel quantity will interact. Previous studies tested if stress affects vowel-quantity perception in such languages; this study explores conversely if vowel quantity affects stress perception in English learners of Czech. Since stress is word-initial in Czech, a word-boundary detection task was used, where stress either fell on a long vowel (stress and vowel length coincided) or on a short vowel, but there was a long vowel distracter in an adjacent syllable (stress and vowel length conflicted). Seventeen L1-English-L2-Czech and 69 Czech listeners were tested. Unexpectedly, natives scored slightly but significantly ($p<.05$) better when stress and vowel length conflicted than when they coincided. The opposite was true for the non-natives who were only at chance when stress and vowel length conflicted. No correlation between the non-native scores and various indices of L2 proficiency was found. It is concluded that learners transferred L1 perceptual strategies to the L2.

4pSC13. Identification and discrimination of tonal pitch in speech and nonspeech stimuli for Chinese- and English-native speakers. Chang Liu (Dept. of Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712)

Given that Chinese is a tonal language while English is not, the present study is to examine how language background affects tonal identification and discrimination in speech and nonspeech signals for Mandarin-Chinese and American-English native speakers. The fundamental frequency (f0) contour was systematically manipulated from tone 2 (rising), to tone 1 (level), and to tone 4 (falling) for three types of signals: Chinese vowels /ɑ/, English vowels /æ/, and tonal sweeps. Both groups of listeners showed categorical perception in the identification of tones 1, 2, and 4, while Chinese-native speakers showed sharper boundaries across tonal categories than English-native speakers. Just-noticeable difference (JND) in tonal pitch was also measured for the standard tone 1, tone 2, and tone 4 of the three types of signals. JND was significantly smaller in speech stimuli for Chinese-native speakers than for English-native speakers, but quite comparable between the two groups of listeners for nonspeech stimuli. There results indicate that tonal pitch may be perceived specifically in speech stimuli, and for nonspeech stimuli, the capacity to process the tonal sweeps is similar across listeners with different language backgrounds.

4pSC14. Tonal confusion patterns in Mandarin by Cantonese listeners. Jung-yueh Tu (Dept. of Linguist., Simon Fraser Univ., 8888 Univ. Dr., V5A1S6 Canada, jta31@sfu.ca)

It is well-attested that linguistic experience affects the perception of non-native sounds. The vast majority of research on L2 perception has been carried out on segmental features from the perspective of phonetic similarity between the L1 and L2 sound systems. The general goal of this study is to expand our understanding of cross-linguistic comparison in non-native speech perception, focusing on the somewhat understudied area of perceptual assimilation at a suprasegmental level. More specifically, it deals with comparisons of prosodic systems in two tone languages, Mandarin and Cantonese. The current study examined perception of Mandarin tones by Cantonese listeners. In Experiment 1, Mandarin tones were presented and the Cantonese listeners were requested to identify which tone they heard. In Experiment 2, the Cantonese listeners were instructed to rate how similar each Mandarin tone was to a Cantonese tone. Preliminary results suggest that tonal confusion errors may result from not only the similar acoustic

properties of the tone pairs but also perceptual assimilation between L1 and L2 tonal contrasts. These findings are discussed in terms of the effects of L1 prosodic system on L2 perception and how perceptually assimilation patterns predict listeners' perception performance at the domain of lexical tones.

4pSC15. Acoustic-phonetic characteristics of naturally-elicited clear speech in British English. Rachel Baker and Valerie Hazan (Speech Hearing and Phonetic Sci., UCL, Chandler House, 2 Wakefield St, London WC1N 1PF, UK, rachel.baker@ucl.ac.uk)

Clear speaking styles are characterized by enhancements of specific acoustic-phonetic aspects of the speech signal. However, most studies of casual and clear speaking styles have been based on read speech recorded in laboratory conditions, both normally and when instructed to speak clearly. In this study, casual and clear speech produced by 40 talkers of southern British English was elicited in unscripted dialogues between two talkers via a series of "spot the difference" picture tasks, based on the "diapix" task developed by Bradlow and collaborators. In the "casual speech" condition, the two talkers conversed normally while doing the task. In the "clear speech" condition, to simulate communication between a normal-hearing talker and a talker with a cochlear implant, one of the talkers heard the other talker's speech passed through a three-channel vocoder. Additionally, talkers read sentences containing specific keywords that also occur in the diapix task, both normally and when instructed to speak clearly. Preliminary acoustic-phonetic analyses of the speech corpus will be presented. The clear speech elicited from the diapix and "read speech" conditions will be compared to ascertain whether clear read speech shows similar characteristics to clear speech elicited in more natural communicative situations. [Work supported by ESRC.]

4pSC16. Korean listeners' perception of consonant clusters in English. Gwanhi Yun (Dept. of English, Daegu Univ., 15 Naeri, Jinryang, Gyeongsan, Gyeongbuk, Korea 712-714, ghyun@daegu.ac.kr)

Perception experiments were conducted to see whether Korean listeners perceive the illusionary vowel in the legitimate sequence of English consonants, especially in onset positions. Korean listeners were presented three types of licit consonant clusters in English (e.g., sC in spee; Cy in kyoo; obstruent + approximant in plee) and had to choose between the original sequence and vowel-repaired sequence. First, Korean listeners did not perceive the illusionary inserted vowel between two consonants with all three types of onset clusters (sC, Cy, OA). Perception of the insertion of unmarked vowel was obtained only 20.1 %, while consonant clusters (disallowed in Korean) were not repaired in perception. Second, perceptual repair with illusive vowels was adopted the most in the sequence of obstruents and approximants (pree; 36 %), next of s+consonant (spee; 20%), and the least of Cy sequences (pyoo; 15%). Our result indicates that loanword phonology might be primarily computed on the basis of the phonological grammar of the borrowing language rather than perceptual assimilation based on phonetic-fine representation [Peperkamp *et al.* (2008)] . Further, it shows that perceptual repair by illusory vowels in the illicit onset clusters are gradual, not categorical unlike phonetic implementation of loanword phonology.

4pSC17. Improved learning of a non-native phonetic contrast by combining active task performance with passive stimulus exposures. Nicole Marrone and Beverly A. Wright (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208, n-marrone @northwestern.edu)

Adults can learn to hear the phonetic distinctions of another language with practice. It is commonly assumed that such improvements require active performance of the target task throughout training. However, we have seen that, for other perceptual tasks, similar performance gains can be achieved by replacing some active-performance trials with passive stimulus exposures. Here we examined the learning of a non-native phonetic contrast using several training protocols that varied in the proportion of active and passive trials during the practice period. Native, monolingual English speakers were trained to identify a Thai phonetic contrast along a voice-onset-time continuum. A group given half active training and half passive exposures showed as much improvement in the category boundary as a group given active training for the entire practice period. The active-passive group out-performed a group given half active training and half training on an unrelated task with no passive exposures. Finally, a group given only passive stimulus exposures showed the least improvement. Results suggest that learning a non-native phonetic contrast via active task performance can be reinforced by passive stimulus exposures. Therefore, active-passive training could be useful in real-world speech training applications. [Work supported by NIH/NIDCD.]

THURSDAY AFTERNOON, 21 MAY 2009 GALLERIA NORTH, 2:00 TO 5:00 P.M.

Session 4pSP

Signal Processing in Acoustics: Pattern Recognition in Acoustic Signal Processing II

Grace A. Clark, Chair

Electronics Engineering, Lawrence Livermore National Lab., Livermore, CA 94550

Chair's Introduction—2:00

Invited Papers

2:05

4pSP1. A tutorial on nonstationarity detection in acoustic signals: Parametric and nonparametric approaches. Patrick J. Wolfe (Statist. and Inf. Sci. Lab., Harvard Univ., Oxford St., Cambridge, MA 02138, patrick@seas.harvard.edu)

This tutorial provides an overview of nonstationarity detection in acoustic signals, focusing on model-based parametric approaches as well as more flexible nonparametric ones. The techniques discussed are presented in the context of speech and audio waveforms, with several real-world examples given, but also apply more broadly to any class of acoustic signals that exhibits locally stationary behavior. Many such waveforms, in particular information-carrying natural sound signals, exhibit a degree of controlled nonstationarity, and are often well modeled as slowly time-varying systems. The tutorial first describes the basic concepts of such systems and their analysis via local Fourier methods. Parametric approaches appropriate for speech are then introduced by way of time-varying linear predictive models, along with nonparametric approaches based on variation of time-localized estimates of the power spectral density of an observed random process. [Work supported in part by DARPA, NGA, and NSF.]

2:45

4pSP2. Bayesian source tracking in an uncertain ocean environment. Stan E. Dosso and Michael J. Wilmut (School of Earth and Ocean Sci., Univ. of Victoria, Victoria B.C. V8W 3P6, Canada)

This paper considers matched-field tracking of a moving acoustic source in the ocean when acoustical properties of the environment (water column and seabed) are poorly known. The goal is not simply to estimate source locations, but to determine localization uncertainty distributions, thereby quantifying the information content of the tracking process, and to use this information to probabilistically predict future source locations. To this end, a Bayesian formulation is applied in which source and environmental parameters are considered random variables constrained by noisy acoustic data and by prior information on parameter values (e.g., physical limits for environmental properties) and on interparameter relationships (limits on source velocity). Source information is extracted from the posterior probability density (PPD) by integrating over unknown environmental parameters to obtain a time-ordered series of joint marginal probability surfaces over source range and depth. Given the strong nonlinearity of the localization problem, marginal PPDs are computed numerically using efficient Markov-chain Monte Carlo methods, including Metropolis-Hastings sampling over environmental parameters (rotated into principal components and applying linearized proposal distributions) and heat-bath Gibbs sampling over source locations. Several approaches to estimating optimal tracks, track uncertainties, and future source locations from the set of marginal-probability surfaces are considered.

3:25

4pSP3. Adaptation of tandem hidden Markov models for non-speech audio event detection. Mark Hasegawa-Johnson, Xiaodan Zhuang, Xi Zhou, Camille Goudeseune, and Thomas Huang (ECE Dept. and Beckman Inst., Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801)

Non-speech audio event detection (AED) could be used for low-cost, spatially diffuse surveillance applications, e.g., monitoring of vehicle activity in a national park, or of footsteps in a hallway. Experiments have shown that non-speech AED benefits from the dynamic inference strategies such as the hidden Markov model (HMM), but that the acoustic features useful for non-speech events may not be the same as those useful for speech. One possible solution is a tandem HMM: an HMM whose observation vector is constructed from the output of an instantaneous discriminative classifier, e.g., a neural network. The use of tandem HMMs for non-speech AED is hindered, however, by the relatively small size of most non-speech-audio training corpora. This talk will demonstrate that tandem HMMs can be trained to detect non-speech audio events using a novel form of regularized training: Baum–Welch back-propagation (as proposed by Bengio *et al.*), using the conjugate-gradient adaptive form of the Baum–Welch auxiliary function (as proposed by Lee *et al.*, and as commonly used in maximum *a posteriori* HMM adaptation).

3:45

4pSP4. Statistical signal characterization in ocean acoustic tomography and geoacoustic inversions. Michael Taroudakis and Kostas Smaragdakis (Dept. of Mathematics, Univ. of Crete and FORTH/IACM, Knossou Ave, 71409 Heraklion, Greece, taroud@math.uoc.gr)

A performance study of a statistical characterization of an underwater acoustic signal in relation to geoacoustic inversion or tomography problems is presented. The method of characterization has been presented by Taroudakis *et al.* [JASA **119**, 1396–1405 (2006)] and is based on the use of an appropriate distribution law which describes the statistics of the sub-band coefficients of the signal wavelet transform. The method has been applied with synthetic data simulating tomographic experiments with low-frequency sources in shallow environments for the estimation of the water column and bottom properties. In this work the inversion procedure incorporating a genetic algorithm is applied in shallow water environments with simulated noisy data to assess the performance of the method and its limitations under realistic conditions.

4:05—4:15 Break

Contributed Papers

4:15

4pSP5. Wavelet-based neural networks applied to automatic detection of road surface conditions using tire noise from vehicles. Wuttiwat Kongrattanaprasert, Hideyuki Nomura, Tomoo Kamakura (Dept. of Elec. Eng., Univ. of Electro-Commun.s, 1-5-1 Chofugaoka, Chofu-shi, Tokyo 182-8585, Japan, wuttiwat@ew3.ee.uec.ac.jp), and Koji Ueda (Nagoya Electric Co., Ltd., Miwa-cho, Ama-gun, 490-1294, Japan)

The detection of road surface conditions is an important process in efficient road management. In particular, in snowy seasons, prior information about the road conditions such as an icy state, helps road users or automobile drivers to obviate serious traffic accidents. This paper proposes a novel approach for automatically detecting the states of the road surface from tire noises of vehicles. The method is based on a wavelet transform analysis, artificial neural networks, and the mathematical theory of evidence. The proposed method employs the wavelet transform using multiresolution signal decomposition techniques. The proposed classification is carried out in sets of multiple neural networks using learning vector quantization networks. The outcomes of the networks are then integrated using the voting decision-making scheme. It seems then feasible to detect passively and readily the states of the surface, i.e., as a rule of thumb, the dry, wet, snowy, and slushy state, automatically.

4:30

4pSP6. Discrimination of blasting sounds, music, and wind noise using a Gaussian mixture-model classifier. D. Keith Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, D.Keith.Wilson@usace.army.mil) and Michael J. White (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL 61822)

Wind noise and acoustic signals were recorded outdoors with a horizontally oriented, square, 7×7 microphone array having an intersensor spacing of 0.914 m. Even with this very small spacing, the wind noise was found to have very little spatial correlation at frequencies above 15 Hz, thus indicating that the turbulent eddies responsible for the wind noise distort very rapidly. This property can be used to distinguish wind noise from acoustic signals possessing similar frequency content. To demonstrate this idea, wavelet processing was applied to recordings with predominantly wind

noise, propane cannon blasts, and unsteady but persistent sounds (music). A set of features involving ratios of the wavelet coefficients at different frequencies and between different microphones was then formulated. Three of the features relate to the time-domain shape of the signals, two to the spectral content, and three to the spatial correlation. A Gaussian-mixture-model classifier was trained with the feature statistics of the three signal types, and classifier predictions were then compared against an independent test data set. Results indicate a 96.6% correct classification rate of the signals. Signal shape features reliably distinguish the blasting and music, whereas the coherence features reliably distinguish the wind noise from acoustic signals.

4:45

4pSP7. Audio enhancement of biomechanical impact data. Joe Guarino, Wes Orme, and Wayne Fischer (Dept. of Mech. and Biomedical Eng., Boise State Univ., Boise, ID 83725-2075, jguarino@boisestate.edu)

Analysis and interpretation of impact data from a force transducer or accelerometer can be augmented and enhanced using audio playback. Trends and differences which may be difficult to identify using data imagery can be elucidated and reinforced by converting digital data from a force plate to an audio signal, which can then be played through a high-quality speaker system. The audio stream can be processed using standard acoustical methods such as tempo and pitch shifting, which can emphasize frequencies and tone bursts for improved signal characterization. We apply audio enhancement to data from two separate biomechanical studies: (1) a drop landing experiment for the investigation of gender differences in impact upon landing and (2) an experiment for the investigation of impact differences between cleated and noncleated shoes on artificial turf. The continuous wavelet transform (CWT) is used to process data from a force plate in both studies. Results are compared using the semblance analysis approach described by Cooper and Cowan ["Comparing time series using wavelet-based semblance analysis," Computers Geosciences 34, 95–102 (2008)]. Interpretation of results is enhanced using speaker-driven audio output synthesized from the CWT and semblance analysis.

THURSDAY AFTERNOON, 21 MAY 2009 — GRAND BALLROOM I, 1:00 TO 4:00 P.M.

Session 4pSW

Speech Workshop

Note: This is the first poster session scheduled for the Cross-Language Speech Workshop. Please see page 2751 for abstracts of the papers to be presented in this session.

THURSDAY AFTERNOON, 21 MAY 2009 — BROADWAY I/II, 2:15 TO 5:30 P.M.

Session 4pUW

Underwater Acoustics and Structural Acoustics and Vibration: Monostatic and Bistatic Detection of Elastic Objects Near Boundaries: Methodologies, and Tradeoffs II

Mario Zampolli, Cochair
NATO Undersea Research Ctr., 19138 La Spezia, Italy

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Invited Papers

2:15

4pUW1. Exact and approximate techniques for scattering from targets embedded in a layered medium. Ahmad T. Abawi and Michael B. Porter (HLS Res., Inc., La Jolla, CA 92037)

To be able to accurately compute scattering from a target embedded in a layered medium (waveguide), the scattering and propagation problems must be solved as a single boundary value problem. This is accomplished by solving the wave equation in an environment that contains both the target and the waveguide and satisfying boundary conditions on the surface of the target and the boundaries of the waveguide. One way that this can be accomplished is by the use of the virtual source technique, which replaces the target with a collection of sources whose amplitudes are determined from the boundary conditions on the surface of the target. This method converts the problem of scattering from a target in a waveguide to a multisource propagation problem. In this paper, the virtual source technique is used to compute scattering from a target in a waveguide and various ways to speed up computation are examined. These include the use of various propagation models to propagate the field produced by the virtual sources to the receiver. Various solutions are compared and the advantages and disadvantages of each model are discussed.

2:35

4pUW2. Bistatic specular reflection by a proud cone: Experiments and interpretation. Jon La Follett and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, lafollej@mail.wsu.edu)

For sufficiently small grazing angles of illumination, specular backscattering of high frequency sound by a proud cone having a vertical symmetry axis is not observed from geometric considerations. Observations from scaled tank experiments suggest, however, that specular scattering from a conical target in this configuration can be observed at shallow grazing angles bistatically. Bright features in the scattering corresponded to sound that had interacted with the flat boundary and the cone and to direct acoustic reflections from the cone. Feature locations were in agreement with a geometric analysis of bistatic reflection [P. Marston, J. Acoust. Soc. Am. **124**, 2584 (2008)] modified so as to allow for images caused by the flat surface. Data were obtained by suspending the apex of a small solid aluminum cone through the air-water interface of a tank. The cone was illuminated from below at grazing incidence and the free surface simulated the ocean floor. Bistatic measurements were obtained by scanning a hydrophone along a line parallel to the interface at varied depths. Data were also obtained from a truncated cone. [Research supported by ONR.]

2:55

4pUW3. Excitation of low frequency modes of solid cylinders by evanescent waves: Mode properties and coupling. Aubrey L. Espana, Philip L. Marston (Dept. of Phys. and Astron., Washington State Univ., Pullman, WA 99164-2814), and Kevin L. Williams (Univ. of Washington, Seattle, WA 98105)

When using sound to detect objects buried beneath the seafloor, situations arise in which the incident acoustic wave has a significant evanescent component. This situation has been simulated in tank experiments [Osterhoudt *et al.*, IEEE J. Ocean. Eng. (in press)] and the simulation was used to investigate scattering mechanisms. In prior work, the backscattering of evanescent and ordinary-propagating waves from small solid aluminum cylinders was studied [Espana and Marston, J. Acoust. Soc. Am. **124**, 2584 (2008)]. It was determined that a strong low frequency mode could be excited when the cylinder was highly tilted in a horizontal plane. With increasing simulated burial depth, the spatial decay rate of the backscattering was enhanced compared to the spatial decay rate of the evanescent soundfield. This resonance has since been driven by evanescent waves when the cylinder is highly tilted in a vertical plane. This alternate method of excitation also showed an enhanced spatial decay rate with increasing simulated burial depth. FEM simulation of the free-field response of the cylinder reveals that this mode causes a rocking motion of each end of the cylinder and it is plausible that evanescent waves could also excite this type of response. [Work supported by ONR.]

3:15—3:30 Break

3:30

4pUW4. Burial depth dependence of the bistatic scattering amplitude for cylinders illuminated by evanescent waves using two–dimensional finite elements. Nicholas R. Cerruti, David B. Thiessen, and Philip L. Marston (Phys. and Astron. Dept., Washington State Univ., Pullman, WA 99164-2814, ncerruti@wsu.edu)

Prior research examined the dependence on simulated burial depth of the low-frequency scattering by small targets illuminated by evanescent waves [P. L. Marston, A. L. Espana, C. F. Osterhoudt, and D. B. Thiessen, J. Acoust. Soc. Am. **122**, 3034 (2007)]. The backscattering amplitude from targets with localized coupling displayed a spatial decay rate approximately twice that of the evanescent wave. An extended reciprocity relation was proposed which accounts for the more general case of a bistatic observation in the water column above the sea floor. In the bistatic case the spatial decay rate may differ from the case of backscattering. The present research concerns the testing of generalized reciprocity for small circular cylinders using two-dimensional finite elements. The calculated spatial decay rate for low-frequency bistatic scattering follows the generalized reciprocity condition when the predicted decay rate (for the specified observation scattering angle) exceeds the spatial decay rate of the incident evanescent wave. This computational result includes agreement with the double decay-rate case of backscattering. The calculations indicate that bistatic observation can significantly reduce the spatial decay rate of the signal dependence on burial depth. [Work supported by ONR.]

3:50

4pUW5. Bistatic and monostatic scattering from elastic structures using boundary element methods in free space and near plane boundaries. Ralf Burgschweiger, Martin Ochmann (Beuth Hochschule fuer Technik Berlin, Luxemburger Str. 10, D-13353 Berlin, Germany, burgi@tfh-berlin.de), and Ingo Schaefer (Forschungsanstalt der Bundeswehr fuer Wasserschall und Geophysik, D-24148 Kiel, Germany)

The bistatic and monostatic numerical calculation of the pressure scattered from structures composed of elastic materials and possibly filled with another material is one of the main purposes for the detection of underwater objects. For this reason, the sound pressure scattered from spherical objects placed in and filled with fluid will be calculated in the frequency domain. The results of an in-house developed BEM-package which supports single and multiple fluid-structure-interactions will be compared with analytical solutions based on spherical wave functions and with results of commercial BEM/FEM applications. Performance optimizations of the calculation process for the uncoupled rigid and coupled monostatic case as a result of using a parallelized matrix creation and solution with a specific variant of the direct solving process will be presented. We will also compare results for a cubic structure, placed in water above a finite plane boundary, with an equivalent half-space solution that incorporates a suitable half-space Green's function and could be used for fast approximations in the mid- and high-frequency range.

4:10

4pUW6. Synthetic aperture sonar simulations of cylindrical targets. Steven G. Kargl, Kevin L. Williams, Eric I. Thorsos (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40 St., Seattle, WA 98105, kargl@apl.washington.edu), and Joseph L. Lopes (Naval Surface Warfare Ctr., Panama City, FL 32407)

Cylindrical targets of finite length can be used as reference targets not only for calibrating an existing SAS system, but more importantly, for testing new classification and identification algorithms. With only a few well-characterized measurements available for proud and buried cylindrical targets, numerical simulations of the acoustic response of these targets offer the potential to realize an unlimited set of target orientations with respect to the source and receiver locations. This paper discusses recent progress with our acoustic scattering models and the generation of a set of pings suitable for SAS processing. SAS images generated from numerical simulations are compared to SAS images generated from data collected during the recent pond experiment 2009 (Pondex09) at NSWC-PCD's facility 383. The target is a solid aluminum cylinder with a 0.3 m diam and length of 0.61 m. [Work sponsored by ONR and SERDP.]

Contributed Papers

4:30

4pUW7. Wavefront curvature and near-field corrections for scattering by targets buried under sea-floor ripple. Raymond Lim, Gary S. Sammelmann, and Joseph L. Lopes (Naval Surface Warfare Ctr. Panama City Division, Panama City, FL 32407)

Curvature of the incident field wavefront has been implicated in discrepancies between modeled and measured pressure fields scattered at shallow grazing angles from targets buried under ripple in NSWCPCD's test pond, where the range to the target is limited to about 10 m. Improvements on our attempts to correct for these and other near-field effects in existing incident-plane-wave-based models is described here. Explicit numerical integration of incident field expansion coefficients as well as the basis functions needed to formulate the transition matrix solution for the scattering process is used in order to create a reliable benchmark for comparisons with simpler corrections and measured data. It is found that some care must be exercised in performing the required two-dimensional wave vector integrals because the ripple wave vector can cause integration to be unstable in the perturbative terms accounting for ripple effects. Results are compared with recent modifications of existing sonar simulations and data collected in NSWCPCD's test pond. [Work supported by ONR and SERDP.]

4:45

4pUW8. Bistatic scattering from underwater unexploded ordinance and the impact of burial. Joseph A. Bucaro (SET, Inc., Naval Res. Lab., 4555 Overlook Ave., Code 7130, Washington, DC 20375-5320, joseph.bucaro.ctr@nrl.navy.mil), Brian H. Houston (Naval Res. Lab., Washington, DC 20375), Larry Kraus (Global Strategies Group (North America), Crofton, MD 21114), Harry J. Simpson, David C. Calvo, and Louis R. Dragonette (Naval Res. Lab., Washington, DC 20375-5320)

Interest in exploring various sonar approaches for detecting underwater unexploded ordnance (UXO) has been growing in large part because of the strong support of the Strategic Environmental Research and Development Program (SERDP). Among the many issues now being explored are the following two fundamental questions: what are the broadband acoustic scattering characteristics associated with typical submerged UXO, and how are these impacted by the bottom sediment? Recently, the authors reported laboratory grade, *monostatic* acoustic scattering measurements on UXO targets in the free field. In the present study, further echo measurements are obtained on UXO both to infer what merits may exist for exploiting *bistatic* echo responses as well as to address the effect of sediment on the target echo characteristics. The measurements, which were carried out in the NRL sediment pool laboratory and free-field facilities, include bistatic *free-field* scattering for three source directions viz. 0°, 45°, and 90°, and bistatic measurements in the sediment facility with the target *proud, half-buried*, and *fully buried*. The data suggest that access to bistatic echo information in operations aimed at detecting submerged UXO-like targets could provide an important capability, especially for buried targets. [Work supported by SERDP and ONR.]

5:00

4pUW9. The broadband in-water structural acoustics of unexploded ordinance: Tank comparisons with at-sea rail measurements. Harry J. Simpson, Brian H. Houston, Mike L. Saniga, Joe A. Bucaro (Naval Res. Lab., Code 7130, 4555 Overlook Ave., Washington, DC 20375-5320), Alain R. Berdoz, and Larry A. Kraus (Global Strategies Group (North America) Inc., Largo, MD 2077)

Free field, proud, and buried laboratory measurements of unexploded ordinance (UXO) are compared to at-sea rail-based measurements of the same UXO. The Laboratory for Structural Acoustics (LSA) at the Naval Research Laboratory consists of a 1 million gallon, deionized water, indoor cylindrical tank (17 m diam by 15 m deep) and an indoor rectangular tank (10 m by 8 m) laboratory, with a 3 m deep sand bottom and 4 m of water column. These pristine laboratory measurements are used to explore the physical acoustics of the sound-structure interactions that can be used to validate models and to understand the structural acoustic features that may be measured in littoral environments. These laboratory results are compared and contrasted with measurements made on the same UXO in St. Andrews Bay near Panama City, Florida using a rail-based synthetic aperture sonar (SAS). The structural acoustics responses of the UXOs in this shallow water, 8 m deep, sandy-mud bottom environment are compared with the pristine results from the LSA. [Work supported by SERDP and ONR.]

5:15

4pUW10. Long-range low frequency shallow water acoustic propagation and bottom penetration experiments for underwater unexploded ordinance. Harry J. Simpson, Mike L. Saniga, Brian H. Houston (Naval Res. Lab., Code 7130, 4555 Overlook Ave., Washington, DC 20375-5320), Alain R. Berdoz, Philip Frank, Steve W. Liskey, and Larry A. Kraus (Global Strategies Group (North America) Inc., Largo, MD 20774)

Among the many issues now being explored in the acoustic detection underwater unexploded ordinance (UXO) in shallow water is the impact of propagation and bottom interaction. Here, we report on long range (1 km) propagation measurements in a littoral environment at the sediment-water interface in the 1 kHz–12 kHz frequency band. The water channel studied was an 8 m deep arm of St. Andrews Bay, Panama City, Florida. The medium grain sandy bottom had a compressional waves peed of 1700 m/s, and the wavespeed in the water was 1539 m/s. A dense array of point pressure measurements (1 cm vertical sampling) were acquired for a synthetic vertical aperture starting from 2 m above and going through 2 m below the interface and at three ranges, 78, 485, and 990 m. The results show deep penetration into the sediment at long ranges for frequencies below 5 kHz. The data are examined using a two-fluid parabolic equation model, and the overall environmental acoustics are discussed including the nature of sediment penetration due to evanescent waves, multipath, and interface roughness.