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
Woods Hole Oceanographic Inst., Woods Hole, MA 02543

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 accurately 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 con-

siderable 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.