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

11:20

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

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

11:35

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

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

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

Session 4pAAa

Architectural Acoustics: Acoustics of Health and Healing Environments

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

Chair's Introduction—1:00

Invited Papers

1:05

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

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

1:25

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

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

1:45

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

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

2:05

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

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

2:25

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

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

Contributed Papers

2:45

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

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

3:00

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

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

3:15

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

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

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

Session 4pAAb

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

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

Chair's Introduction—2:00

Invited Papers

2:05

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

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

2:30

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

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

2:55

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

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

3:20—3:35 Break

3:35

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

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

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

4:00

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

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

Contributed Papers

4:25

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

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

4:40

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

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

Session 4pAB

Animal Bioacoustics: General Topics in Animal Bioacoustics II

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

Contributed Papers

3:00

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

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

3:15

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

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

3:30

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

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

3:45

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

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

4:00

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

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

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

4:15

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

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

4:30

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

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

4:45

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

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

Session 4pBB

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

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

Chair's Introduction—1:15

Invited Papers

1:20

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

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

1:40

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

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

2:00

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

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

2:20

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

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

2:40

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

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

3:00—3:20 Break

3:20

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

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

3:40

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

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

4:00

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

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

4:15

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

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

4:30

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

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

4:45

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

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

5:00

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

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

Session 4pNSa

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

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

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

Chair's Introduction—1:00

Invited Papers

1:05

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

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

1:25

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

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

1:45

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

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

Contributed Paper

2:05

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

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

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

Session 4pNSb

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

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

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

Chair's Introduction—2:30

Invited Papers

2:35

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

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

2:55

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

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

3:15

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

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

3:35

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

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

Contributed Papers

3:55

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

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

4:10

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

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

4:25—5:10 Panel Discussion

Session 4pPA

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

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

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

Contributed Papers

1:20

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

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

1:35

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

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

1:50

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

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

2:05

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

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

2:20

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

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

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

2:35

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

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

2:50

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

Range compensation functions are applied to echo signals so that the corresponding range-compensated echo intensity equals the volume backscattering coefficient to within a multiplicative constant. For echo integration in a transducer far field, the transmit and receive pressure fields each decrease inversely with range r, and the well-known range compensation function, applied to intensity, is $r^2\times10^{\alpha r/5}$, where α is the absorption coefficient. For echo integration in a transducer near field, the range compensation function is that of the far field modified by a multiplying factor. This factor is, approximately, the ratio of the square of the surface integral of the field intensity at r due to extrapolation from the far field, assuming inverse range dependence of the amplitude, to the product of the surface integrals of transmit and receive intensities at r. This general range compensation function is developed for the special case of echo integration with a parametric sonar in which scatterers are ensonified in a region where the nonlinearly generated difference-frequency field is growing and echoes are received in the far field of a collocated, linearly operating transducer. It is evaluated numerically for the Kongsberg TOPAS PS18 parametric subbottom profiling sonar, with primary-frequency band 15–21 kHz and difference-frequency band 0.5–6 kHz.

3:05

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

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

3:20—3:40 Break

Invited Papers

3:40

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

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

4:00

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

The fundamental nonlinear interaction which leads to the scattering of waves and to the end fire array yields a Boltzmann equation for sound, a one fluid theory of superfluid helium [S. Putterman, P. H. Roberts, Physics Letters, **89A**, 444 (1982)] and a route to the

quantum theory of interacting phonons [M. Cabot, S. Putterman, Phys. Lett. **83A**, 91 (1981)]. Along with a review of these consequences of nonlinear classical acoustics the conundrum of the consistency of the second law of thermodynamics with the Hamiltonian nature of the end fire array will be exposed but not resolved [S. Putterman, P. H. Roberts, Phys. Rep., **168**, #4 (1988)].

Contributed Paper

4:20

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

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

Invited Paper

4:35

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

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

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

Session 4pPP

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

Kathryn H. Arehart, Chair

Dept. of Speech, Language and Hearing Science, Univ. of Colorado at Boulder, Boulder, CO 80309

Contributed Papers

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Session 4pSAa

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

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

Invited Paper

1:25

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

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

Contributed Papers

1:50

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

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

2:05

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

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

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

Session 4pSAb

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

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

Invited Papers

2:25

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

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

2:50

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

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

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

3:15

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

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

Contributed Paper

3:40

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

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

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

Session 4pSC

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

Megha Sundara, Chair

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

Contributed Papers

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and contributors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

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

Session 4pSP

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

Grace A. Clark, Chair
Electronics Engineering, Lawrence Livermore National Lab., Livermore, CA 94550

Chair's Introduction—2:00

Invited Papers

2:05

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

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

2:45

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

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

3:25

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

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

3:45

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

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

4:05—4:15 Break

Contributed Papers

4:15

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

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

4:30

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

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

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

4:45

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

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

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

Session 4pSW

Speech Workshop

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

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

Session 4pUW

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

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

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

Invited Papers

2:15

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

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

2:35

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

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

2:55

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

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

3:15—3:30 Break

3:30

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

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

3:50

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

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

4:10

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

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

Contributed Papers

4:30

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

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

4:45

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

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

5:00

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

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

5:15

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

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