the intrinsic spectral properties of emitted signals to discriminate intruder emission from acoustic noise sources such as ship traffic typically present in the river. Further heuristic processing exploits the periodic nature of signals such as human breathing, by detecting the beginnings and endings of the diver's inhalation signals. The detector assesses whether the recognized features are consistent with previously documented breathing rates in order to automatically and robustly detect the presence of a diver. The developed approach was applied in diver detection tests conducted in the Hudson River estuary where cross-correlation technique allowed finding the line of bearing to a diver. The challenges to implementation of such method in a standalone acoustic buoy under development at Stevens are also discussed. [This work was supported by ONR Project No. N00014-05-1-0632: Navy Force Protection Technology Assessment Project].

11:30

4aUW13. Finite difference time-domain simulation of Scholte wave generation and propagation along a shallow waterway bottom containing an anomaly. Thomas Muir (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, One Coliseum Dr., University, MS 38677, tmuir@olemiss.edu) and Dwynn Lafleur (Univ. of Louisiana at Lafayette, Lafayette, LA 70504)

Results are presented on a two-dimensional simulation of the generation and propagation of a Scholte wave pulse along a water-sediment interface in a shallow water waveguide, consisting of 12.5 m of water, overlying a like depth of sand sediment. An underwater sound source, 2.5 m above the sediment, generates a low-frequency Ricker wavelet, which propagates as "modal tone bursts" in the waveguide and the sediment. Some of the energy is partitioned into a propagating Scholte wave mode at the sediment interface, a planar surface containing an "anomaly," i.e. a moundlike feature. Acoustic leakage into the elastic sediment is demonstrated, as is the propagation of the Scholte wave impulse moving along the interface and traveling over the anomaly, at a much slower velocity than its acoustic mode counterpart in the water column. The results are presented as a movie sequence, composed of snapshots of the spatial amplitude distribution of the Scholte wave as it propagates. Also presented are the modeled seismometer signals, yielding a two-dimensional particle velocity (hodographs) in the vertical plane at the anomaly, and somewhat beyond it. The significance of low-frequency Scholte waves in the acoustics of extremely shallow water is discussed.

11:45

4aUW14. A model of distant shipping noise. Cathy Ann Clark, Randall T. May, and Kristy A. Moore (NUWC-DIVNPT, 1176 Howell St., Newport, RI 02841)

Shipping noise from high-density areas is believed to migrate down the coastal shelf and propagate to long ranges via the deep sound channel, resulting in an increase in noise level at low angles for submerged receivers. A propagation calculation, which is applicable when a high degree of cycle mixing results in nearly random summation of path effects, is introduced and used to predict low-frequency (1–300 Hz), long range shipping noise. The vertical directionality of the noise is computed by fitting a source distribution within the deep sound channel by the method of least squares. Comparisons to a limited number of low-frequency measured data sets are presented.

THURSDAY AFTERNOON, 13 NOVEMBER 2008 LEGENDS 9, 1:30 TO 4:25 P.M.

Session 4pAA

Architectural Acoustics: Innovative Integration of Acoustic Treatment into Modern Architecture

Scott D. Pfeiffer, Cochair
Threshold Acoustics LLC, 53 W. Jackson Blvd., Ste. 1734, Chicago, IL 60604

Molly K. Norris, Cochair
Threshold Acoustics LLC, 53 W. Jackson Blvd., Ste. 1734, Chicago, IL 60604

Chair's Introduction—1:30

Invited Papers

1:35

4pAA1. Development of special acoustic materials for the Experimental Media and Performing Arts Center at Rensselaer Polytechnic Institute: Part 1. Carl Giegold (Threshold Acoust., 53 W. Jackson Blvd., Ste. 1734, Chicago, IL 60604, cgiegold @thresholdacoustics.com)

A series of project-specific interior acoustic elements were developed for the eMPAC project. Most prominent is the fabric ceiling in the concert hall, which was developed in tandem with the canopy of a similar material recently installed in London's Royal Festival Hall. Other concert hall finishes were also the subject of much mathematical and physical study during the design process. In the two studios, the finishes respond to the client's acoustic metaphor of a forest clearing, where the acoustic response is relatively alive but highly diffuse at mild and high frequencies. A system of GFRG panels, each of which has several different scales of shaping, was devised and empirically tested in several iterations to achieve the desired result.

1:55

4pAA2. Development of special acoustic materials for the Experimental Media and Performing Arts Center at Rensselaer Polytechnic Institute: Part 2. Zackery Belanger (Kirkegaard Assoc., 8th Fl., 801 W. Adams St., Chicago, IL 60607, zbelanger @kirkegaard.com)

Modern facilities for experimental performance create rigorous acoustical challenges because of the extreme and peripheral nature of their use. Spaces firmly rooted in tradition have acoustic criteria, which are largely known, but experimental spaces must accommodate the traditional, the contemporary, and the not yet conceived. The Experimental Media and Performing Arts Center at Rensselaer Polytechnic Institute, opening this year in Troy, NY, is one such facility. Five major venues—a concert hall, a visual performance studio, a musical performance studio, a theater, and a production suite—are implemented in a complementary harmony of flexibility. The concert hall includes a lightweight Nomex ceiling, which selectively reflects and transmits sound, and the studios utilize adjustable diffusers and absorbers of custom design. Isolation of the critical venues is extensive in anticipation of the use of extreme levels and low frequency sound.

2:15

4pAA3. Case study: Exposed structure acoustical solutions. Kenneth Good and Sean Browne (Armstrong W. I., 2500 Columbia Ave., Lancaster, PA, 17601)

Many new buildings employ an exposed structure design aesthetic and traditional construction elements such as ceilings and walls are being eliminated. These materials perform an acoustical function that is sacrificed in order to achieve visual and sustainability requirements. This presentation will explore several problematic exposed structure spaces and the acoustical solutions that restored proper performance.

2:35

4pAA4. Is that "acoustic?" Unexpected finishes used to shape the acoustic environment. Gregory Miller, Evelyn Way, Byron Harrison, and Richard Talaske (TALASKE, 1033 South Boulevard, Oak Park, IL 60302)

The visual aesthetic desired by many contemporary architects includes surfaces that are incompatible with acoustic requirements for sound absorption and sound diffusion—particularly concave walls and large expanses of smooth, flat surfaces. Additionally, the necessity of sustainable design requires the creative use (and reuse) of unexpected construction materials. Case studies to be presented include the Jewish Reconstructionist Congregation of Evanston, Illinois, the Arena Stage in Washington, DC, The Harman Center for the Arts in Washington, DC, the Steppenwolf Theater in Chicago, Illinois, and the North Campus Auditorium at the Walgreen Drama Center of the University of Michigan at Ann Arbor. These projects will be used to highlight the use of woven textures, shaped precast, and poured-in-place concrete, screens, and repurposed construction waste to solve common (and some uncommon) acoustic challenges.

Contributed Papers

2:55

4pAA5. Stretched, perforated, and auralized: A case study of integrating acoustic treatment into the modern museum to control sound propagation and room acoustics. Ryan Biziorek (Arup, 155 Ave. of the Americas, New York, NY 10013)

Modern museum designs typically require exhibits with AV components. The museum spaces must also be flexible for various types of internal and external events. While the current architectural trend to use glass, concrete, and other sound reflecting materials presents acoustic challenges, there are various techniques to integrate a sound absorbing treatment into a museum facility to allow for flexibility and control of room acoustics. Using materials integrated into multifunction architectural elements, acoustic treatment can be seen as a necessary element within a space. Further modeling and auralization can help the client understand the amount and necessity of an acoustic treatment within various museum spaces. Two case studies will be presented as to how acoustic treatment can be incorporated into the architectural design and use auralization to allow the client to be involved in the acoustic design process.

3:10

4pAA6. Acoustical optimization of shapes and materials used in modern architecture. Peter D'Antonio (RPG Diffusor Systems, Inc., 651-C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com)

Classic architecture or classically inspired architecture benefits from the fact that scattering surface, in the form of columns, statuary, and relief ornamentation, is an integral part of the architecture. As architecture evolved into using less ornate surfaces in smooth rectilinear and more recently curvilinear forms, this created a need to design scattering surfaces that complement modern architecture. When contemporary scattering surfaces are required, shape optimization, using any reflective material, has proven to be mutually useful for the acoustician and the architect. A shape drawn by the architect is parametrized and then optimized maintaining the desired motif. An iterative computer program that combines the benefits of boundary element and multidimensional minimization techniques will be described and examples will be shown. When absorption is required, microperforation and microslit designs offer a novel solution, using light transmitting plastics and metals. Microperforated wood veneer panels are also available in which the perforations are barely visible at normal viewing distances. A summary of the theory and application of these ideas can be found in work of Cox and D'Antonio [*Acoustic Absorbers and Diffusers: Theory, Design and Application* (Spon, 2004)].

3:25—3:40 Break

3:40

4pAA7. Modified modal analysis for damped enclosures. Buye Xu and Scott Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT 84602, buye.xu@gmail.com)

Modal analysis (MA) has been widely used to model and understand acoustic fields in undamped or lightly damped enclosures. When additional damping is introduced, MA is known to fail in predicting acoustic quantities close to the boundaries and to exhibit a much slower speed of convergence. Modifications have been introduced into a standard MA approach to partially match the damped boundary conditions and source conditions. Notable improvements in both accuracy and convergence speed are observed when using this modified modal analysis. The theoretical derivation of the model will be discussed briefly, and several examples will be shown to demonstrate the improved accuracy of the model.

3:55

4pAA8. Mitigating low-frequency sound with tunable acoustic chimneys. Bonnie Schnitta and Roy Freeedman (SoundSense LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

A way to absorb and filter low-frequency sound that is generated by sound sources in one or more enclosures is described. A sound generating room is encapsulated within an outer acoustic absorbing room that contains various openings to several acoustic ducts. The acoustic ducts function as an acoustic waveguide to the open air. At each end of the acoustic duct is an impedance matching tapered acoustic horn. In the sound generating room the horn serves as an acoustic absorber that tapers sound into the acoustic duct. At the opposite end, the horn serves as an acoustic radiator of sound energy to the open air. Both radiator and absorbers may be tuned to transmit frequencies past a certain cutoff. The effect is similar to a pressure release valve. The acoustic waveguides may be coupled to tunable passive or active noise mitigation devices such as Helmholtz resonators and low-frequency noise sources that utilize phase information.

4:10

4pAA9. Innovative green acoustic techniques to control noise ingress and sound propagation and to reduce energy consumption. Ryan Biziorek (Arup, 155 Ave. of the Americas, New York, NY 10013)

The current LEED rating system currently does not give credit for any elements of acoustic design within the point system. Too often, projects can become too focused on obtaining the points within the defined categories while sacrificing the basic elements of acoustic design (i.e., noise ingress from natural ventilation). The acoustic consultant must often find innovative ways to incorporate acoustic design into the building that will support the sustainability goals of the building and can be used to support and/or obtain a LEED point. Three innovative methods of green acoustic design in project case studies will be presented: natural ventilation techniques to maintain privacy, acoustic comfort in indoor working environments, and energy saving techniques for performing arts facilities. If applied to a project in an innovative method, the acoustics of a project need not be compromised and could be used as a LEED credit for innovation in design.

THURSDAY AFTERNOON, 13 NOVEMBER 2008 — LEGENDS 2, 1:30 TO 4:00 P.M.

Session 4pAB

Animal Bioacoustics: General Topics

David C. Swanson, Chair

Pennsylvania State Univ., Applied Research Lab., P.O. Box 30, State College, PA 16804

Contributed Papers

1:30

4pAB1. Frequency overrepresentation found in the inferior colliculus of Japanese house bats, *Pipistrellus abramus*. Kazuhiro Goto and Shizuko Hiryu (Life and Medical Sci., Doshisha Univ., 1-3 Tatara Miyakotani, Kyotanabe 610-0321, Japan)

In this study, we examined the best frequencies of the inferior collicular neurons of Japanese house bats, *Pipistrellus abramus*. The bats emit long (>10 ms) and shallow FM pulses (quasi-CF) during their search phase, narrowing the frequency range at around 40 kHz. If the narrowed frequency range is important for detecting a small frequency change or frequency modulation, the bats may need a finer frequency resolution than other frequency ranges. So, the neurons responsive to the frequency range at around 40 kHz must be over-represented somewhere in the auditory system, typically in the inferior colliculus. In order to examine this hypothesis, the tonotopic organization of the inferior colliculus of *P. abramus* was measured electrophysiologically by metal microelectrodes. Results show that the frequency range corresponding to their quasi-CF frequency, at around 40 kHz, appears to be over-represented in the inferior colliculus. [Research supported by an ONR grant.]

1:45

4pAB2. Biosonar features extracted from natural frequencies for natural landmark classification. Jinsong Dai, Maosen Wang (School of Mech. Eng., Nanjing Univ. of Sci. & Tech., Nanjing, 210094, People's Republic of China, smewms@yahoo.com), and Rolf Mueller (Shandong Univ., People's Republic of China)

To navigate in a natural environment, it is practical for an artificial agent to use vegetation as landmark. This is the way bats have done this for its long history. For an outdoor biosonar system based on digital signal processing, it is ideal to extract characteristic features directly from natural landmarks. Any physical structure can be modeled as a number of springs, masses, and dampers. Vegetation with trunks, stems, and foliage in open air always sways in a certain degree; it is possible to extract features from a series of characteristic natural frequencies with electrical biosonar system. In this work, a biosonar system is utilized to study features extracted from several plants for mobile agent navigation. Experimental results indicate that there is always enough air turbulence in the outdoor environment and these biosonar features are efficient in classifying plants. Besides using natural frequency features with the same wind condition, a laboring calibration should be made to improve this method robustness in different wind environments. [Work supported by NSFC.]

2:00

4pAB3. Spatial unmasking of pure tones in broadband noise by zebra finches (*Taeniopygia guttata*). Micheal Dent and Jarrod Cone (Dept. of Psych., Univ. at Buffalo, SUNY, Buffalo, NY 14260, mdent@buffalo.edu)

Detecting a signal embedded in noise is known to be enhanced by spatially separating the signal and noise in humans and other animals. This process is known as spatial unmasking and is a part of the larger phenomenon of the cocktail party problem. The exact mechanisms of unmasking are unknown, but binaural processes are thought to be at least partially involved. Most animals that exhibit unmasking are fairly adept at localizing pure tones in space. We wished to study spatial unmasking in an animal that is very poor at sound localization: the zebra finch. Zebra finches were trained using operant conditioning techniques and the psychophysical method of constant stimuli to peck keys for food reinforcement when they detected a tone embedded in a broadband noise masker. Thresholds were obtained for pure tones ranging from 500 Hz to 4000 Hz when the signal and the noise were emitted from the same speaker and when they were emitted from speaker locations separated by 180 deg. Zebra finches showed relatively little unmasking and there was large variation across subjects and frequencies, suggesting that the mechanisms underlying sound localization are related to those that result in spatial unmasking.

2:15

4pAB4. Categorization of budgerigar (*Melopsittacus undulatus*) warble elements. Hsiao-Wei Tu, Edward Smith, and Robert Dooling (Dept. of Psych., Univ. of Maryland, College Park, MD 20742)

The warble song of budgerigars is composed of a variety of elements without any obvious sequential order. Some of the elements also occur as single utterances. A previous study classified warble elements into 42 groups by visual inspection of spectrograms. However, the density of warble (about 140 elements/min, up to 30 min in duration) makes this method both laborious and time-consuming for analyzing a large amount of warble. Here three human raters took 860 elements from 3 birds and sorted the sonograms into 9 general groups with an inter-rater reliability of 89%. Next, these elements were used to train a neural network. This network learned to categorize a large number of warble elements efficiently with 84% reliability (compared to human raters). Further examination of other warble streams revealed that warble elements are not evenly distributed across these nine groups for the same bird, but the relative proportion of different elements in warble categories is similar across three budgerigars. Ongoing studies are examining whether birds vary the proportion of elements in different social contexts to better understand the biological function of this complex vocalization. [This work is supported in part by DC-00046 and DC-00198 to R.D.]

2:30

4pAB5. The effect of social and environmental manipulations on the vocal response rate of budgerigars (*Melopsittacus undulatus*). Peter Marvit (Dept. of Psych., Univ. of Maryland, College Park, MD 20742, pmarvit@gmail.com), Zach Payne (Univ. of Maryland, College Park, MD 20742), Adam Ratner (Bates College, Lewiston, ME 04240), and Robert Dooling (Univ. of Maryland, College Park, MD 20742)

The budgerigar (parakeet) exhibits complex social interactions and relationships. This project investigates the budgerigar “contact call” that is used for both local and distance social cohesion. It is hypothesized that budgerigar vocal response rates will vary with the social “distance” of the caller (e.g., mates greater than conspecific strangers) and that overall rates will increase as the background acoustic environment becomes more complex and naturally rich. To test these hypotheses, vocal response rates of budgerigars were measured to four groups of recorded contact calls, within four acoustic environmental contexts. Three mated pairs were subjects. The results showed huge individual differences. Overall, males responded significantly more than females. There was a trend toward greater responses to mates than self, flockmates, or strangers. There were a large and significant response increases during a familiar flock background and significant decreases during silence. The response vocalizations themselves were analyzed for possible patterns to the different auditory contexts. The results are consistent with a hypothesis of social facilitation, although the individual differences pose a curious problem that a larger sample size may help elucidate. [Work supported by NIH/NIDCD 5R01DC000198 and 2P30DC004664.]

2:45

4pAB6. Transmission fidelity in rhesus monkey (*Macaca mulatta*) “coos” and “screams”. Eric Tarkington, Lisa Heimbauer, and Michael Owren (Dept. of Psych., Georgia State Univ., P.O. Box 5010, Atlanta, GA 30302-5010, etarking@ooadvocate.com)

In animal communication, studies of environmental acoustics examine issues such as habitat-specific propagation effects on vocalizations and likely impact on psychologically significant aspects (e.g., localizability and identifiability). The current work compared transmission fidelity in harmonically structured “coos” versus noisy “screams” produced by female rhesus monkeys (*Macaca mulatta*). Calls were broadcast in a temperate, mixed forest using a loudspeaker (Genelec 1029A) positioned either 0.5 m or 1.0 m above the ground, and re-recorded using two microphones (Sennheiser MKH106T) and a digital audio deck (TAS-CAM DA-P1). The “near” microphone remained at a constant distance of 2.5 m from the loudspeaker, while the “far” microphone was positioned at 10, 20, 30, 40, or 50 m. Eight different coos and eight different screams were played four times each at every distance, with cross-correlation values calculated for each pair of near- and far-microphone recordings. Results included that correlations decreased monotonically with far-microphone distance and were higher for calls broadcast at 1.0 m than at 0.5 m. Furthermore, however, correlations were significantly higher for coos than for screams, consistent with other evidence that harmonically structured calls are better suited to functions such as long-distance transmission of caller identity than are noise-based vocalizations.

3:00

4pAB7. Ultrasonic characterization of insect wing reflectivity and wing-beat motion at 200 kHz. David Swanson (Appl. Res. Lab., Penn State Univ., State College, PA 16804), Tom Baker and Ryan terMeulen (Acoust. Program, Pennsylvania State Univ., State College, PA 16802)

A bistatic active ultrasonic sensor is used to detect and identify an insect attracted to a pheromone lure in the beam intersection. The ultrasound beam is a continuous waveform (CW) of approximately 200 kHz to avoid detection by the insect auditory system, which is very sensitive near 40 kHz and below to detect predatory bats. Both amplitude modulation (AM) and frequency modulation (FM) of the CW beam are examined. The reflection factor of the wing material is measured using ultrasonic pulses. The near-field interference nulls of the transmitting beam are modeled and measured to ensure that the lure is not placed in one of these zones. While good AM and FM signals are seen in the received waveform, the pivoting and flapping motions of the wings along with the insect movements create a complex pulse-train like waveform. This is seen to be inherently due to the narrow beam-width for the transducers and a lack of nonspecular scattering from the moving wings. The data show that a simple AM detector can be used to detect the presence of an insect at the lure and to measure the wing-beat frequency. [Work supported by USDA].

3:15

4pAB8. Numerical analysis of flapping sound generated by hovering insects. Yoshinobu Inada (Numerical Anal. Group, Aerosp. Res. and Development Directorate, Japan Aerosp. Exploration Agency (JAXA), 7-44-1, Jindaiji-Higashimachi, Chofu, Tokyo 182-8522, Japan, inada@chofu.jaxa.jp), Hikaru Aono (Univ. of Michigan, Ann Arbor, MI 48109), Hao Liu (Chiba Univ., Chiba 263-8522, Japan), and Takashi Aoyama (Japan Aerosp. Exploration Agency (JAXA), Chofu, Tokyo 182-8522, Japan)

Insect flapping sound is a consequence of flow disturbance generated by flapping wing motion. Spatial and temporal changes of pressures on the wing surface and vortex structures generated by the wing motion are considerable sources of the flapping sound. To analyze such mechanism in the sound generation, we have developed an integrated method combining computational fluid dynamics (CFD) techniques and acoustic analysis. Unsteady flows around a hovering insect are simulated by NS solution-based CFD analysis with a multiblocked, moving-overset grid technique. In the acoustic analysis, monopole and dipole sound sources generated by the wing motion are analyzed by using Ffowcs–Williams and Hawkings method. In this study, numerical analyses of the flapping sounds are carried out for three insects of hawkmoth, honeybee, and fruitfly. The CFD analysis provides a detailed picture of the flow fields around the hovering flyers where complicated vortex aspects and the induced pressure distributions on the wing surfaces show concrete evidence of the sound sources. The acoustic analysis further clarifies the characteristics of the flapping sounds induced by these sound sources, including the sound directivity and its spectrum distribution. These results effectively demonstrate the cause-and-effect relationship in sound generation and thus indicate the availability of this integrated CFD-acoustic method.

3:30

4pAB9. Creating a regionally focused online archive for natural sounds: The Western Soundscape Archive. Jeff Rice (J. Willard Marriott Library, Univ. of Utah, 295 South 1500 East, UT 84112, jeff.rice@utah.edu)

Variations in species distribution and abundance, general topography, as well as regional dialects among species ensure that no two places or environments will sound the same. Regionally focused environmental sound archives can play an important role in documenting these variations. This pa-

per will describe the recent creation of the Western Soundscape Archive (Westernsoundscape.org), an ongoing project led by the J. Willard Marriott Library at the University of Utah that seeks to build a representative and free online resource of animal and environmental sounds of 11 contiguous western United States. The archives primary includes the region's terrestrial vertebrates, as well as targeted ambient recordings. By gathering existing sound recordings, creating new recordings where appropriate, and employing innovative mapping technology, the archive hopes to document and preserve the soundscape of the West and to create a replicable model for other libraries in other regions. [Work supported by an IMLS National Leadership Grant.]

3:45

4pAB10. Relative saliency of envelope and fine structure cues in zebra finch song. Beth Vernaleo and Robert Dooling (Neurosci. and Cognit. Sci. Program, Dept. of Psych., Univ. of Maryland, College Park, MD 20742, bgoldman@umd.edu)

Birdsong provides a useful model for communication and vocal development, and zebra finch song in particular is attractive for its acoustical complexity and repetitive nature. Males sing one song for the purpose of mating display and territory defense, whereas females do not sing. In this study, we are particularly interested in which acoustic features of a male's song are most perceptually salient. Using a repeating background of a single song motif, zebra finches were trained to discriminate changes to two cues in song: increases in intersyllable interval duration (envelope cue) and reversals of single syllables within a song motif (fine structure cue). Results show that zebra finches are able to discriminate changes to fine structure of syllables much more easily than changes to the overall envelope of song, specifically intersyllable intervals. Further experiments have been done using a noise burst modeled song in which song syllables were replaced by frozen random noise bursts of the same duration. Results show that zebra finches are able to discriminate single noise burst reversals, suggesting that they are able to attend to and follow fine structure on a very small time scale. [Work supported by NIH/NIDCD 5R01DC000198 and 2P30DC004664.]

THURSDAY AFTERNOON, 13 NOVEMBER 2008 LEGENDS 3, 1:30 TO 4:20 P.M.

Session 4pEA

Engineering Acoustics, ASA Committee on Standards and Biomedical Ultrasound/Bioresponse to Vibration: High Precision Acoustical Measurements

Victor Nedzelnitsky, Chair
Natl. Inst. of Standards and Technology, 100 Bureau Dr., Gaithersburg, MD 20899-8220

Invited Papers

1:30

4pEA1. A radiation force technique for acoustic power calibration of high intensity focused ultrasound transducers. Subha Maruvada (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO62-2222, Silver Spring, MD 20993)

It is essential to know the acoustic power radiated by transducers used in high intensity focused ultrasound (HIFU) surgery devices for both safety and effectiveness considerations. The power radiated by medical ultrasound transducers usually is measured via radiation force balance (RFB) methods. However, for the high power focused fields encountered in HIFU applications, such measurements can be difficult due to the need for short measurement times to prevent transducer damage, heating of the RFB target, and bubble formation. To address these challenges, a procedure based on pulsed measurements was formulated. High output focused ultrasound transducers were characterized in terms of an effective duty factor, which was then used to calculate the power during the pulse at high drive levels. Two absorbing target designs were used, and both gave reliable results and displayed no damage and minimal temperature rise if placed near the HIFU transducer and away from the focus. The procedure was reproducible up to the maximum power generated of approximately 230 W, representative of the HIFU range, thus allowing the radiated power to be calibrated in terms of the peak-to-peak voltage applied to the transducer.

1:50

4pEA2. Reducing measurement errors: Taking the design of microphones to the limit. Johan Gramtorp and Erling Sandermann Olsen (Brüel & Kjær Sound and Vib. Measurements A/S, Skodsborgvej 307, DK-2850 Nærum, Denmark)

It is impossible to make one microphone that covers all possible applications. In the real world, the properties of a microphone are the result of a number of carefully chosen compromises based on the anticipated measurement situation and the measurement equipment. The electroacoustic properties of condenser measurement microphones are rigidly bound to the physical properties of the microphone cartridge and the electrical properties of the preamplifier. This means that in the design of a measurement microphone the challenge is to find the right balance between the various physical properties of the microphone. However, recent development in simulation models, construction technology, and knowledge of material properties has opened for the design of microphones that has pushed the limits. A new class of general purpose measurement microphones has been born. In this paper, the physical properties that determine the electroacoustic properties of a condenser microphone are reviewed, and the various compromises that must be chosen are discussed. The possibilities and limitations in microphone design are illustrated with examples of the newest microphone designs from Brüel & Kjær.

2:10

4pEA3. Phase frequency responses of measurement microphones and their calibration. Erling Frederiksen (Danish Primary Lab. of Acoust. and Brüel Kjaer, Skodsborgvej 307, 2850 Naerum, Denmark)

For more than 20 years, comparison calibration of microphone phase responses has been a topic of interest. The interest grew up in the 1980s, when it became technically possible to design instruments for measurement of sound intensity and particle velocity. At that time, the calibration frequency range was limited by the first generation of comparison couplers, which typically worked up to 5 kHz. Since then, new applications and needs for wider frequency range or higher accuracy have occurred. Today phase response comparison calibration of half-inch and quarter-inch microphones can be made with couplers from below 20 Hz up to about 20 kHz. The uncertainty is estimated to about 0.010 deg between 20 and 500 Hz from where it increases to some tenths of a degree at 20 kHz. Microphone array applications require large numbers of microphones with essentially equal phase responses. This has increased the need for reference standards, whose absolute phase responses are known. The Danish Primary Laboratory of Acoustics calibrates such reference standards by the reciprocity method. This paper gives an overview of the spread of microphone phase responses, commonly required tolerances, phase calibration methods, and their uncertainties.

2:30

4pEA4. Acoustical impedance measurement. Charles King (Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60143)

Many components of specialty transducers have extremely tight acoustic impedance tolerances and become difficult to measure in a production environment. Comparing the device to the acoustical impedance of a cavity is one method of determining its acoustical impedance. A precise method of determining the acoustical impedance is described. The technique applies methods normally found in defining cavities for pressure reciprocity calibration to obtain a precision acoustical reference. Pressure reciprocity standards also define how environmental factors influence acoustical properties and these are included into the system to maintain the precision regardless of the environmental conditions or the frequency of operation. Further improvements are obtained by employing differential measurements throughout the measurement and calibration process to eliminate many common errors common to field calibration of microphones. Ultimately the measured data are curve fitted to an equivalent circuit model to extract the desired components from the measurement artifacts. This talk will discuss the method of defining the acoustical components and extracting the desired value from the equivalent circuit model.

2:50

4pEA5. Developments at the National Institute of Standards and Technology in pressure calibration of laboratory standard microphones. Victor Nedzelnitsky, Randall P. Wagner, and Steven E. Fick (Natl. Inst. of Standards and Technol., 100 Bureau Dr., Stop 8220, Gaithersburg, MD 20899-8220, victor.nedzelnitsky@nist.gov)

Improvements in apparatus and procedures for determining the pressure sensitivities of IEC Types LS1Pn and LS2aP laboratory standard microphones by the reciprocity method aim at reducing the effects of significant components of uncertainty. One such component involves the uncertainty with which the front cavity depths of the microphones are known. Therefore, various noncontact methods and procedures for measuring these depths using a depth-measuring microscope have been developed and investigated. Other significant uncertainty components involve the voltage measurements performed during the reciprocity method. Careful consideration of the dependence of measurement uncertainty on voltmeter operational mode, voltmeter range, signal voltage variation with frequency, and the voltages at which range switching occurs can significantly reduce the uncertainty components involving voltage measurements. Another component, involving the uncertainty with which the temperature of the gas in an acoustical coupler is known, can contribute significantly to the uncertainty with which the acoustic transfer impedance in the reciprocity method can be determined. Measurement of acoustical coupler temperature during calibration can significantly reduce this component. Examples of how improvements aimed at reducing these uncertainty components are being systematically incorporated in an evolving test bed are described.

3:10

4pEA6. Primary sinusoidal calibration of accelerometers. Lixue Wu and Peter Hanes (Inst. for Natl. Measurement Standards, Natl. Res. Council, Bldg. M-36, Ottawa, ON K1A 0R6, Canada, lixue.wu@nrc-cnrc.gc.ca)

Accurate vibration measurements require suitably calibrated transducers. A new system for primary sinusoidal calibration of the voltage sensitivity of accelerometers according to Method 1 (fringe-counting) of ISO 16063-11 has been implemented at the NRC Institute for National Measurement Standards. The system includes a new shaker that is capable of generating sufficient acceleration for calibration of accelerometers at frequencies as low as 2 Hz. The design, construction, and testing of the system will be described. Factors that influence the measurement uncertainty of the calibrations such as transverse motion, rocking motion, acceleration stability, distortion, and repeatability have been evaluated and will be discussed.

3:30

4pEA7. Microphone calibration. Gunnar Rasmussen (G.R.A.S. Sound Vib., Skovlytoften 33, 2840 Holte, Denmark, gras@gras.dk)

Microphone calibration accurate acoustical measurements are desirable for legal reasons and for research and development. The uncertainty in acoustic measurements may cause uncertainty in determining improvement in design and development of products. International standards may still be improved for calibration of the measurement technique at low frequencies and at high frequencies especially. The absolute calibration of measurement microphones is dominated by reciprocity calibration. This has been dominating in acoustic metrology for many years and rightly so for the middle frequency range 100–1000 Hz. Other methods are very useful at low

frequencies and at high frequencies with a good overlap to the reciprocity technique and may offer lower uncertainty especially for calibration in the field where the major part of acoustical measurements take place. Microphone comparison, piston phone calibration, scaling in microphone size at high frequencies, electrostatic actuator, and directionality are some of the techniques used in practical calibration work.

Contributed Paper

3:50

4pEA8. Frequency limitations of coupler-based calibrations of the pressure sensitivity of measurement microphones. Allan J. Zuckerwar (Analytical Services and Mater., 107 Res. Dr., Hampton, VA 23666) and Qamar A. Shams (NASA Langley Res. Ctr., Hampton, VA 23681)

Calibration of the pressure sensitivity of measurement microphones in a coupler requires by definition that the incident pressure be uniform over the surface of the microphone diaphragm. This requirement is readily satisfied at frequencies up to the cut in frequencies of the radial modes and, if the microphone is not azimuthally symmetric, the spin modes. Above the lowest natural frequency of either of these modes, the uniformity of the pressure distribution will be destroyed. This effect is alluded to in current standards, ANSI S1.15-2005/Part 2 and IEC 61094-5:2001, but the suggested countermeasure "to use more than one coupler with different dimensions" or to compare "calibrations performed in a variety of other jigs and couplers" does not resolve the problem. Specification of the maximum excitation frequency to preserve pressure uniformity in a coupler is presented for several laboratory standard and working standard microphones.

4:05

4pEA9. Demonstration of an impedance based method for sonar calibration and monitoring. Corey Bachand, David A. Brown, and Boris Aronov (BTech Acoust, LLC and Electro-Acoust. Res. Lab., ATMC/ECE, UMass Dartmouth, 151 Martine St., Fall River, MA 02723)

It is well known that impedance based methods can be used to monitor the performance degradation of sonar transducers. A new approach that involves precise measurements of the voltage (*V*), current (*I*) and phase (*P*), termed the *VIP*-impedance method, has been developed and implemented by the authors for characterizing individual transducer and array performance in sonar systems. Examples of predicted transducer sensitivities and beam patterns on both cylindrical and tonpiltz sonar arrays using the new method are presented and compared with experimental data.

THURSDAY AFTERNOON, 13 NOVEMBER 2008 LEGENDS 10, 1:30 TO 3:20 P.M.

Session 4pED

Education in Acoustics and ASA Student Council: Project Listen Up

James M. Sabatier, Chair
Univ. of Mississippi, Natl. Ctr. for Physical Acoustics, University, MS 38611

Chair's Introduction—1:30

Invited Papers

1:35

4pED1. A Helmholtz resonator experiment for the Listen Up project. Chad A. Greene, Theodore F. Argo, IV, and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78712-0292)

The Listen Up project seeks to develop an educational booklet and apparatus that will be used to foster interest in acoustics and teach basic acoustical concepts to middle school students. The packaged kit must be low cost; hence this experiment uses a 16 oz plastic water bottle as a Helmholtz resonator, a disposable syringe, and an inexpensive pitch pipe. Understanding the Helmholtz resonator is fundamental to many applications in acoustics. A simple algebraic model of the device, derived from a mechanical mass-spring analogy, relates the volume of air in the bottle to the resonance frequency of the system, and hence the pitch that is produced when one blows over the bottle opening. The volume of air inside the bottle can be easily and accurately controlled by adding water with a graduated syringe. The dependence of pitch on the volume of air in the bottle can be systematically demonstrated, and the validity of the model can be checked by comparing the sound produced to musical notes from the pitch pipe. This experiment yields a physical understanding of a common acoustical phenomenon and inspires further interest and study of more complex phenomena.

1:55

4pED2. Standing waves on a string and electromagnetic induction. Joseph Gladden, III (Dept. of Phys., Univ. of Mississippi, University, MS 38677, jgladden@phy.olemiss.edu)

Standing waves on a string are a popular classroom demonstration which helps students understand such fundamental acoustic concepts as wavelengths, interference, and resonances. The close connection between this demonstration and stringed musical instruments helps bridge the gap between the science classroom and other interests they may have. In this version of the demonstration, we replace the string with a copper wire and add a permanent magnet. This allows the vibrations in the string to be directly translated into an electrical signal which can be broadcast through a standard stereo amplifier and speakers similar to an electric guitar pickup. The

effects of string tension and length can then be explored. Alternatively, an ac can be driven through the wire creating a periodic driving force from the magnetic field of a precise frequency allowing standing waves to be established on the string. More advanced students can also examine the Fourier transform of the signal produced when the wire is struck and find peaks at the resonant frequencies.

2:15

4pED3. Tuning forks, resonators, and beats: Demonstrations for "Project Listen Up". Murray S. Korman (Dept. of Phys., U.S. Naval Acad., 572 C Holloway Rd., Annapolis, MD 21402, korman@usna.edu)

Gearing to an audience of all ages, a demonstration apparatus is presented including student versions of Hermann von Helmholtz's tuning fork sounder, circa 1859 (where an intermittent current in an electromagnetic coil drives the tuning fork, which is located near a resonator), and Alexander Graham Bell's tuning fork experiment, circa 1876 (where a variable resistance circuit due to a slight tuning fork contact in a conductive liquid causes a relay to vibrate at the same frequency as the fork). Using the popularity of a basic mechanical acoustical apparatus and briefly unfolding the histories behind Helmholtz combining several of his tuning fork sounders to generate vowel sounds and Bell's first telephone, students are then given careful explanations of the demonstrations. Next, the focus shifts to well developed basic experiments that students can perform in a laboratory setting with emphasis placed on presenting the details of a carefully thought out scientific approach. "Project Listen Up" (an initiative of the Education in Acoustics Committee) hopes to develop an educational apparatus for a broad range of learners as one of their goals. [Work supported by USNA.]

Contributed Papers

2:35

4pED4. Introducing perceptual coding using a double-blind A/B testing demonstration. Scott Porter (The Graduate Program in Acoust., The Penn State Univ., 217 Appl. Sci. Bldg., State College, PA 16802, scott.porter@psu.edu)

Many of today's college students experience music in compressed formats on a regular basis. While most are familiar with the term "MP3," only a few know of the psychoacoustic principles and perceptual coding schemes which this and other compressed music formats rely upon. In fact, it is not uncommon that they do not realize the audible differences that exist between the CD-quality audio files and those compressed at common commercial bit rates. To illustrate this, the author used freely available software (WINABX and AUDACITY) to demonstrate the differences with double-blind A/B testing as part of an introductory course in acoustics offered at the Pennsylvania State University. The software will be briefly explained, the questionnaire used to acquire the students' responses will be shown, and an interactive demonstration will be included.

2:50

4pED5. Outdoor pulse echo speed of sound measurement for all ages. James M. Sabatier (NCPA, Univ. of Mississippi, University, MS 38655), Charles H. Sabatier (Fairfax County Public Schools, Alexandria, VA 22309), and Celeste S. Taylor (Putnam City Schools, Oklahoma City, OK 73162)

One of the major thrusts of ASA Vision 2010 is to develop education outreach for a spectrum of learners from kindergarten students to senior citizens. Along these lines we explore a simple speed of sound measurement that can be accomplished by this broad range of learners. Understanding slow speed of sound compared to the speed of light allows the learner to comprehend the lack of temporal/optical correlation between sound and position of airplanes as they move across the sky. Other examples include the late arrival of sounds from aerial salutes during firework displays and sounds of lightning during storms. Synchronizing the acoustic echo of a sound pulse from an outdoor wall with the pulsed source for several duty cycles allows sound speed measurements. Analysis of the time of arrival with distance is presented for various wall sizes and air temperatures. The successes of the learning exercise and results of the measurements when accomplished by the K-gray audience will be presented (see, for example, http://arts.ucsc.edu/EMS/Music/tech_background/TE-01/soundSpeed.html).

3:05

4pED6. Visualization of harmonic starting process for flute sound experiment. Yoshinori Takahashi (Dept. of Comput. Sci., Kogakuin Univ., Tokyo, Japan, yoshinori@ieee.org), Mikio Tohyama (Waseda Univ., Tokyo, Japan), and Kazunori Miyoshi, Prof. (Kogakuin Univ., Tokyo, Japan)

Signal analysis based on spectral accumulation has great potential for monitoring the condition of structures using random vibration records excited by natural forces. This article describes cumulative harmonic analysis (CHA) by introducing a spectral accumulation function into Berman and Fincham's conventional cumulative analysis, thus exploring a potential new area in cumulative analysis. CHA effectively visualizes the feedback process from the flow of air against an edge. In this work, we visualized the harmonic starting process for understanding of physical experiment of musical instrument study especially the flute.

Session 4pMU

Musical Acoustics, Speech Communication, and Signal Processing in Acoustics: Statistical Approaches for Analysis of Music and Speech Audio Signals

Paris Smaragdis, Cochair
Adobe Systems, Inc., 275 Grove St., Newton, MA 02466

George Tzanetakis, Cochair
Univ. of Victoria, Dept. of Computer Sci., P.O. Box 3055, Victoria, British Columbia, V8W 3P6 Canada

Chair's Introduction—1:30

Invited Papers

1:35

4pMU1. Audio analysis using sparse representations. Mark D. Plumbley, Samer A. Abdallah, Maria G. Jafari, and Andrew Nesbit (Dept. of Electron. Eng., Queen Mary Univ. of London, Mile End Rd., London E1 4NS, UK, mark.plumbley@elec.qmul.ac.uk)

The method of "sparse representations," based on the idea that observations should be represented by only a few items chosen from a large number of possible items, has emerged recently as an interesting approach to the analysis of images and audio. New theoretical advances and practical algorithms mean that the sparse representations approach is becoming a potentially powerful signal processing and analysis method. Some of the key concepts in sparse representations will be introduced, including algorithms to find sparse representations of data. An overview of some applications of sparse representations in audio will be described, including for automatic music transcription and audio source separation, and pointers will be given for possible future directions in this area. [This work has been supported by grants and studentships from the UK Engineering and Physical Sciences Research Council.]

1:55

4pMU2. Modeling local stationarity in speech wave forms. Daniel Rudoy, Prabahan Basu, and Patrick J. Wolfe (Statistics and Information Sci. Lab., Harvard Univ., Oxford St., Cambridge, MA 02138)

Typical speech wave forms are well modeled as slowly time-varying, or so-called locally stationary, stochastic processes. This talk outlines recent work in detecting and modeling locally stationary speech time series, and describes a new method of adaptive short-time Fourier analysis and reconstruction based on local measures of time-frequency concentration. While adaptive analysis measures have previously been proposed in order to overcome the limitations of fixed-resolution schemes, the scheme presented here derives from quantifiable and rigorous notions of local stationarity. This yields demonstrable robustness properties for the case of noisy speech, as well as improved mean-square error estimation properties and other quality improvements relative to standard estimation procedures in which time-frequency resolution is fixed. [Work supported in part by DARPA and NSF.]

2:15

4pMU3. Toward automatic music transcription from audio input. Shigeki Sagayama, Hirokazu Kameoka, and Haruto Takeda (Grad. School of Info. Sci. Tech., Univ. of Tokyo, Bunkyo-ku, Tokyo 113-8656, Japan, sagayama@hil.t.u-tokyo.ac.jp)

Transcribing audio input to obtain a music score has been a basic but yet hard problem in music signal and information processing. It can be paralleled with continuous speech recognition comprising of acoustic analysis, acoustic model, language model, and decoder. This presentation discusses functional modules for the music case, i.e., estimation of multiple fundamental frequencies, rhythm models, and chord modeling. For multi-F_0 estimation, a computational auditory scene analysis-motivated approach is taken to model human hearing where each acoustic object is modeled with Gaussian mixture both along frequency and time axis [Kameoka (2005)]. An extended EM-algorithm is applied to iterative estimation of fundamental frequencies and onset/offset timings of music notes contained in the given spectrogram of the audio input. As for rhythm estimation, HMM is used to model the sequence of note lengths where tempo is treated in the model as a varying hidden variable [Takeda (2006)]. Chord progression is also modeled with HMM [Kawakami (2000)] where transition probabilities between chords and emission probabilities of notes from the hypothesized chord have been trained with a music database. Related issues are also discussed including other approaches to multiple fundamental frequencies with specmurt analysis [Sagayama (2005)], non-negative matrix factorization [Raczynski (2007)], and timber modeling [Miyamoto (2007)].

2:35

4pMU4. Prior structures for non-negative matrix factorization based audio source separation. Tuomas Virtanen (Dept. of Signal Processing., Tampere Univ. of Technol., P.O. Box 553, FI-33101 Tampere, Finland, tuomas.virtanen@tut.fi) and Ali Taylan Cemgil (Univ. of Cambridge, Trumpington St., CB2 1PZ Cambridge, UK, atc27@cam.ac.uk)

A generative signal model corresponding to non-negative matrix factorization algorithms is presented, and the model is extended by using priors for the matrices to be estimated. In the analysis of audio signals, this approach allows using prior information about the sounds, while allowing adaptation of the model parameters to the exact characteristics of the observed signal. Specifically, conjugate priors lead to fast inference algorithms. Methods for learning the parameters of the prior distributions from training material are described. The presented approach is evaluated on various sound source separation tasks, including standard data sets of music and speech.

Contributed Papers

2:55

4pMU5. Neighborhood indicator based on tonality and arrangements in musical organization: Automatic system for selecting similar excerpts based on musical information. Yuichiro Yamakaji (Graduate School of Sci. and Technol., Ryukoku Univ., 1-5 Yokoya, Oe-cho, Seta, Otsu-shi, Shiga 520-2194, Japan) and Masanobu Miura (Ryukoku Univ., Japan)

Recently, various methods for searching similar musical excerpts have been studied. However, it has not yet been realized that a method of automatically selecting similar excerpts based on features was obtained from musical information in consideration of a music theory. Therefore, this study was to realize the method. Concretely, employed parameters are "chord trigram," modeling sequence of chords based on the *n*-gram model, "emotional parameter," corresponding to emotional expressions such as tempi, modes, and average of duration for melody, and "note density," corresponding to the occurrence frequency of notes. Excerpts dealt with here are grouped into several categories based on proposed parameters by the CART algorithm, and degree of similarity is calculated between excerpts in each category. Developed here is a system for automatically selecting similar excerpts based on features obtained from musical information, named "NITAMO," which stands for neighborhood indicator based on tonality and arrangements in musical organization. The new system deals with more than 5400 excerpts and employs the XF format, a commercially sold format as an extended form of MIDI. We confirmed that the proposed system was useful by comparing similar musical excerpts provided by proposed system with those by human. [Work supported by the HRC, Ryukoku Univ.]

3:10—3:25 Break

Invited Papers

3:25

4pMU6. Inferring missing spectral data. Paris Smaragdis (Adobe Systems Inc., 275 Grove St., Newton, MA 02466) and Bhiksha Raj (MERL, Cambridge, MA 02139)

In this talk we will present a methodology that allows us to infer missing portions of spectrograms. We will present an approach that constructs models of sounds by observing either examples or the existing portions of a spectrogram with missing data. Once the model is learned we can use it to reconstruct missing areas of a spectrogram with inaudible artifacts. This process is very useful when trying to correct errors from spectral editing or when dealing with processes that corrupt a signal in the time/frequency domain. We will show how this approach is appropriate for polyphonic audio signals and that it significantly outperforms approaches using generic statistical models which are ill suited for spectral data.

3:45

4pMU7. Discovery of temporal patterns in continuous nonrandom sound sequences. Rita Singh and Bhiksha Raj (Human Lang. Technologies Ctr. of Excellence, Johns Hopkins Univ., Stieff Bldg./810 Wyman Park Dr., Baltimore, MD 21211)

The problem addressed is that of automatically determining the minimal structures in structured sounds such as human speech. It is well established that human speech comprises consistent sound units (such as phonemes), a fact that is exploited by speech recognition systems, which only model the units, characterizing all speech as sequences of units. Currently, the units are typically manually defined, and their statistical models are assigned structures based on subjective judgments. In this work it is attempted to identify these units automatically through an analysis of data. The problem is treated as one of entropy minimization. The minimum entropy estimation principle assumes a structured universe and dictates the estimation of the most predictable model that the observations used to train the model will allow. For the current problem, this amounts to identifying a set of units such that every word can be represented by a minimum-perplexity network over them. Experiments demonstrate that this procedure identifies units with consistent structures that, although not always identical to phonemes, are nevertheless able to explain the data just as well as quantified by recognition accuracy. More interestingly, the algorithm is generic and may be employed to learn component units of other natural sounds besides speech.

4:05

4pMU8. AudioDB: Scalable approximate nearest-neighbor search with automatic radius-bounded indexing. Michael Casey (Dept. of Music, Dartmouth College, HB 6242 Hallgarten Hall, Hanover, NH 03755)

This talk describes new approximate nearest-neighbor methods employed in a scalable audio-feature database system called "AudioDB." This open-source system is designed to scale to storing and searching hundreds of millions of feature vectors on standard UNIX workstation platforms. A radius-bounded nearest-neighbor vector-sequence search algorithm, based on locality sensitive hashing (LSH), achieves sublinear retrieval times at this scale. The performance of the LSH-based algorithm depends critically on the choice of radius bound supplied—the wrong value impacts retrieval accuracy or retrieval time. An optimal radius estimator is derived by modeling the minimum value distribution of a random sample of a data set's pairwise distance distribution. When used with LSH this yields

accurate search results with retrieval times several orders of magnitude faster than exhaustive search methods and space-partitioning methods. The same statistical sampling method is used to perform retrieval tasks at successively higher levels of specificity on labeled or unlabeled audio collections. The result is a system that (a) unifies audio retrieval tasks across a range of specificities, using the statistical framework of background distance-distribution sampling and hypothesis testing (b) is as accurate as exhaustive search methods and (c) is three orders of magnitude faster than exhaustive search methods.

4:25

4pMU9. Combining prior-knowledge and grouping cues using a spectral clustering approach. George Tzanetakis (Dept. of Comput. Sci., Univ. of Victoria, Victoria, BC V8W 3P6, Canada, gtzan@cs.uvic.ca) and Luis Gustavo MartinsI (INESC Porto, Portugal)

Learning happens at the boundary interactions beween prior knowledge and incoming data. The same interplay takes place when trying to analyze and separate complex mixtures of sound sources such as music. Many approaches to this problem can be broadly categorized as either model based or grouping based. Although it is known that our perceptual system utilizes both of these types of processing, building such systems computationally has been challenging. As a result most existing systems either rely on prior source models or are solely based on grouping cues. In this work it is argued that formulating this integration problem as clustering based on similarities between time-frequency atoms provides an expressive but discipined approach to building sound source characterization and separation systems and to evaluating their performance. After describing the main components of such an architecture, we describe a concrete realization that is based on spectral clustering of a sinusoidal representation. We show how this approach can be used to model both traditional grouping cues such as frequency and amplitude continuity as well as other types of information and prior knowledge such as stereo panning, onsets, harmonicity, and timbre models for specific integration will also be described.

Contributed Paper

4:45

4pMU10. A theoretical frequency band limitation for analog recordings. Michael Zucker (ARL-Penn State Acoust., Appl. Sci. Bldg., North Atherton St., State College, PA 16802, mxz174@psu.edu)

Proponents of analog recording techniques have described their superiority over digitally sampled signals in part by referencing a theoretically infinite frequency band response. This is a typical argument used in favoring audiophile grade phonograph records to digitally sampled music. The actual frequency response of any individual transducer cartridge or phonograph record is determined by empirical measurements and is often bounded by electrical noise considerations as well as the physical condition of the record being played. These measured and predicted frequency responses, however, still hold the contention that there is theoretically infinite frequency band content available on perfect analog recordings and that the actual limit on frequency content is simply a result of an imperfect implementation of recording and playback. In this study an equation analogous to the Nyquist sampling theorem is derived using the geometry of a phonograph pickup system and simple calculus to obtain an absolute theoretical frequency limit for analog recordings based on the size of the pickup and the speed at which the time signal is written and read. This equation may prove useful in designing analog recording systems specific to a desired frequency band response.

THURSDAY AFTERNOON, 13 NOVEMBER 2008 LEGENDS 4, 2:30 TO 4:15 P.M.

Session 4pNS

Noise and Architectural Acoustics: Sound Levels and Acoustical Characteristics of Modular Classrooms

Paul D. Schomer, Chair
Schomer and Associates, 2117 Robert Dr., Champaign, IL 61821

Chair's Introduction—2:30

Invited Papers

2:35

4pNS1. Sound field amplification competes with noise control. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514)

Sound field amplifiers continue to be aggressively promoted for mainstream classrooms. Studies showing improved academic achievement with sound field systems mislead by not revealing that the subject classrooms were excessively noisy. Sound field is a band-aid solution for noisy classrooms. Surely, quiet classrooms and natural, unstrained voices improve academic achievement as well as or better than amplifiers. Soundfield is being promoted to school building officials as low cost substitutes for noise control. Marketers mislead school officials by suggesting that classrooms cannot be quieted affordably and that occupied classrooms are so noisy that amplification is indispensable. One might ask how students were classroom educated prior to the invention of electronic amplifiers! In a recent "Trojan horse" strategy, sound field systems are being purchased for classroom multimedia. Sound field reinforcement then becomes a "freebie." The problem is that all amplified systems work best in quiet rooms. Moreover, though soundfield amplification works well in special education classrooms for hearing impaired students. It is seldom needed in quiet mainstream classrooms. Moreover hearing impaired students in mainstream classrooms are better served by FM systems. Classrooms that depend on amplification are unavailable when the system becomes inoperative due to malfunction, missing parts, or power failures. Quiet classrooms are robust.

2:55

4pNS2. Progress update from S12 working group 46: Relocatable classrooms. Tom Hardiman (944 Glenwood Station, Ln. Ste. 204, Charlottesville, VA 22901, tom@modular.org)

The purpose of forming working group 46 was to develop an amendment to ANSI S12.60-2002 that specifically focuses on the challenges and unique circumstances surrounding the design, construction, use, and relocation of relocatable classrooms. S12/WG46 is currently drafting an addendum to S12.60-2002 Guideline for Classroom Acoustics. This presentation will provide an update on the progress being made by S12 working group 46.

3:15

4pNS3. Measurement of the ambient in relocateable modular classrooms. Paul Schomer, Wilbur Chang (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821), and Irvin Derks (Bard Manufacturing Co., Bryan, OH 43506-0607)

The ambient in a classroom consists of noises internal to the classroom (e.g., HVAC, lighting, IT equipment, refrigerators, and fish tank pumps) and noises external to the classroom (e.g., street traffic, aircraft, children in the playground, and adjacent classroom noise). Currently, ANSI S12.60, Classroom Acoustics, requires that the A-weighted ambient level, unoccupied, be less than 35 dB at the noisiest position in the classroom. A basic reasonable procedure to implement the required measurements has been developed and tested. The authors find that the procedures developed can be readily used by inexperience but technically oriented individuals. The procedure separates the measurement of internal and external noises into two measurements, typically at different positions. Because the two noises are variable, unrelated in position, and uncorrelated in time, it does not make sense to combine them in any fashion. Rather, in the procedure to be recommended by WG 68, the two measurements are to be reported and evaluated, separately, against the 35 dB criterion. Other notable changes are expansion of the measurement hours to include busy traffic periods (high environmental noise periods) prior to and after school and on Saturdays, and implementation of HVAC duty cycles to better estimate a typical A-weighted hourly equivalent level from HVAC equipment (the subject of a companion paper).

3:35

4pNS4. Outdoor to indoor A-weighted sound level reduction of typical modular classrooms and assessment of potential performance improvements based on the outdoor-indoor transmission class spectrum. Noral Stewart (Stewart Acoust. Consultants, 7406 L. Chapel Hill Rd., Raleigh, NC 27607, asamiami@sacnc.com)

Several different designs for modular classrooms from various parts of the United States were examined for their expected A-weighted outdoor to indoor sound level reduction based on the sound spectrum used for the ASTM E1332 Outdoor Indoor Transmission Class (OITC). Analysis was simplified by using an average absorption effect and the OITC rating of walls and roofs as estimated using the INSUL computer program. Results showed a significant variation among designs with the better designs providing a good foundation for further improvement.

3:55

4pNS5. Exploration of alternative means to evaluate exterior noise loss. Kenneth Good (Acoust. Privacy Enterprises, LLC, P.O. Box 252, Mount Joy, PA 17552)

The outdoor-indoor transmission class measurement can be problematic to execute due to many environmental and logistical issues. This leads to the following question: Can the measurement be made from the inside to the outside? This presentation will explore a technique to evaluate a building's ability to block exterior sound by placing the source signal inside and measurement device on outside.

Session 4pSAa

Structural Acoustics and Vibration: Aeroacoustic and Hydrodynamic Interactions with Structural Acoustics and Vibration

Jerry H. Ginsberg, Chair
Georgia Inst. of Tech., Sch. of Mechanical Eng., Atlanta, GA 30332-0405

Invited Papers

1:30

4pSAa1. Comparison of time-integration schemes for fluid-structure interaction. Aldo A. Ferri, Mohammed Kapacee, Jerry H. Ginsberg (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Tech., Atlanta, GA 30332-0405, al.ferri@me.gatech.edu), and Marilyn Smith (Georgia Inst. of Technol., Atlanta, GA 30332-0150)

Computational structural-dynamics codes invariably use finite element (FE) methods and relatively course meshes, whereas finite-difference methods with fine meshes are popular for modeling the fluid domain. Fluid-structure interaction (FSI) problems in which there is a mean flow feature can be addressed by simultaneously employing both techniques. Because of the dissimilarities of the two formulations, a time-domain solution is most readily obtained by allowing each code to march forward in time in a loosely coupled manner. A proper FSI implementation must address the physical and computational issues associated with compatibility of displacements and surface tractions at the boundary of the two domains. It is also necessary that one identifies a suitable numerical scheme to perform unsteady time-marching simulations of the coupled system. It is the latter issue that concerns this paper. Various techniques for accurate and stable time integration of loosely coupled systems are compared. A two-dimensional example is studied that consists of an inviscid compressible fluid and a thin elastic beam/plate. Small amplitude motion is assumed for the plate, so linear approximations are used in the FE model. The fluid domain, on the other hand, is modeled using the nonlinear Euler equations. [Work supported by NASA Contract No. NAS1-02117, Task Order 6101-GT.]

1:55

4pSAa2. Flight systems aeroelastic-acoustics simulation. Kajal K. Gupta (NASA Dryden Flight Res. Ctr., Edwards, CA 93523), Sangbum Choi (CSULA, Los Angeles, CA 90032), and Adem Ibrahim (Norfolk State Univ., Norfolk, VA 23504)

Many practical problems, such as flight vehicles, are often characterized by complex interactions among a number of primary disciplines as structures, fluids, controls, and propulsion, among others. In some critical flight regimes as transonic flow, the dynamics behavior of fluids tend to be highly complex, needing computational fluid dynamics (CFD)-based modeling, rather than linear aerodynamic methods, for accurate prediction of unsteady flow. Such unsteady pressure values, generated by the interaction of fluids and elastic structures, could then be conveniently used to compute acoustic wave frequencies and sound levels. For accurate simulation of complex engineering problems such as advanced aircraft, it is necessary to employ unstructured grids to model the fluid discipline, such being mostly the case for structure modeling. Effective CFD-based aeroelastic modeling, using finite element discretization of both fluids and structures, employing unstructured grids, have been efficiently utilized for simulation of Hyper-X and F-18 aircraft. In connection with a current ongoing project Stratospheric Observatory For Infrared Astronomy, evaluation of acoustic activities in the cavity of the modified Boeing 747SP aircraft, housing the telescope, became a crucial issue. The multidisciplinary code STARS7 was recently extended to include the acoustic effects.

Contributed Papers

2:20

4pSAa3. Turbulent boundary layer shear stress transmitted through a viscoelastic layer. E. Capone and William K. Bonness (Appl. Res. Lab., Pennsylvania State Univ., P.O. Box 30, State College, PA 16804)

Transfer functions are developed for the transmission of unsteady shear stress, generated by a turbulent boundary layer in water, through a viscoelastic layer backed by a rigid plate. Existing analytical models are used to estimate the unsteady wall pressure and shear stress from 10–1000 Hz for a flat plate boundary layer with zero pressure gradient. A new model is developed for the transmission of unsteady shear stress through the viscoelastic layer. The model is used to predict the unsteady pressure fluctuations, or flow noise (due to the unsteady shear stress), which would be seen by a finite size sensor embedded under the elastomer layer. The calculated unsteady pressure and shear stress levels are in good agreement with recent experimental measurements. The unsteady shear stress transfer functions are found to have a peak at the acoustic wavenumber.

2:35

4pSAa4. Removing unwanted signals from wall pressure and vibration measurements on a structure subjected to turbulent boundary layer excitation. William Bonness, David Jenkins, and Dean Capone (Appl. Res. Lab, Penn State Univ., State College, PA 16804)

Fluid-structure interaction experiments typically involve measurements of the excitation force (or pressure) and the corresponding vibration response to the excitation. In addition to desired flow information, measured turbulent boundary layer wall pressure data often include unwanted signals such as acoustic pressures and vibration induced pressures. Measured vibration data on a structure can also include unwanted electrical noise and vibration energy from adjacent structures. A noise removal technique is presented, which allows one to remove an unlimited number of unwanted correlated signals from a set of measured data. In its simplest form, this technique is related to the coherent output power. However, the more general technique provides an ability to remove multiple signals and to retain complex values (magnitude and phase). These advantages can yield signifi-

cantly greater information of the flow field and structure under investigation. This technique is demonstrated using measurements from an aluminum cylinder internally filled with water flowing at 20 ft/s.

2:50

4pSAa5. A theoretical model to simulate two-phase vibroacoustical frequency and damping effects in a pipe. Vincent Debut and Jose Antunes (Appl. Dynamic Lab., Inst. of Nuclear Technol., Sacavem, Portugal)

In order to avoid excessive flow-induced vibrations in industrial components operating in two-phase flow, the analysis and understanding of energy dissipation mechanisms are of prime importance. Several experimental studies since the classic work [L.N. Carlucci, "Damping and hydrodynamic mass of a cylinder in simulated two-phase flow," J. Mech. Des. **102**, 597–602 (1980)] revealed the strong dependence of two-phase damping on the characteristics of the two-phase flow, particularly the void fraction and fluid used. All these investigations agreed in the complexity of the dissipation processes involved. To tackle such an intricate problem, we consider in this paper the fluid-structure dynamics of a pipe filled with a bubbly liquid which interacts with two single-degree of freedom piston terminations. Several formulations for the coupled problem are stated and compared, the two-phase acoustics being based on a homogeneous mixture formulation [L. van Wijngaarden, "One-dimensional of liquids containing small gas bubbles," Annu. Rev. Fluid. Mech. **4**, 369–396 (1972)]. Then, extensive numerical computations are performed. Plots of the evolution of the modal frequencies and damping as a function of the mixture void-fraction are obtained, and the corresponding vibroacoustical transfer functions and numerical time-domain simulations are presented.

THURSDAY AFTERNOON, 13 NOVEMBER 2008 LEGENDS 1, 3:15 TO 5:30 P.M.

Session 4pSAb

Structural Acoustics and Vibration: Vibration—Measurements and Analysis

Joseph M. Cuschieri, Cochair
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Contributed Papers

3:15

4pSAb1. Performance comparison of viscoelastic materials typically used for acoustical damping of aircraft fuselage structures. Esen Cintosun, Noureddine Atalla (Groupe d'Acoustique de l'Univ. de Sherbrooke, Dept. of Mech. Eng., Univ. de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada), Tatjana Stecenko (MTI Polyfab Inc., Mississauga, ON L5T 2A4, Canada), and Maxime Bolduc (Univ. de Sherbrooke, Sherbrooke, QC J1K 2R1, Canada)

Vibration tests were performed to compare typical viscoelastic damping materials attached to two distinct representative aircraft fuselage structures. The fuselage structures were a 19×20-in^2 Al ribbed panel and a 20×21-in^2 carbon composite ribbed panel. A shaker was used to excite each panel with and without viscoelastic material treatment. Each test structure along with the shaker was situated in an environmental chamber to collect measurements at low temperatures of -20, -30, and -40 °C. A laser vibrometer was used to collect velocity measurements at 15–26 random locations on the surface of the panels. The parameters that were compared included input mobility, damping loss factor (DLF), and space averaged squared velocity. The DLF values that were obtained using the decay rate, power input, and half-power bandwidth methods were presented as averages in 1/3 octave frequency bandwidths (from 100 to 2500 Hz). The space and frequency averaged squared velocity was used to quantitatively categorize the viscoelastic damping materials. The effect of viscoelastic damping material coverage was also evaluated by comparing measurements collected at 50% and 80% coverages. The results of this experimental study are being utilized in finite element modeling, optimization of viscoelastic material treatment, and development of alternatives to viscoelastic damping. [Work supported by the MTI Polyfab Inc.]

3:30

4pSAb2. Coupling of extensional and flexural displacement fields in straight bars. Jerry H. Ginsberg (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, jerry.ginsberg@me.gatech.edu)

Extensional and flexural displacements of straight elastic bars are conventionally treated as independent superposable effects. However, coupling of displacement fields due to boundary effects is an important feature in many realistic configurations. The present paper examines the vibration modes of a beam in which the aforementioned coupling is caused by a roller support that is mounted on an incline. A closed form solution of the coupled dislacement modes is derived. The transcendental characteristic equation is shown to depend solely on the incline angle for the end support, the reduced frequency formed from the bar wave speed, and the ratio of the cross-section's radius of gyration to the span length. The characteristic equation is solved numerically. The resulting eigenvalues are used to evaluate the coupled mode shapes, which leads to a quantitative assessment of coupling effects relative to the limiting cases where the incline is horizontal or vertical, for which the displacement fields truly are uncoupled. An interesting aspect of the analysis is its mathematical similarity to the application of Timoshenko theory to bars.

3:45

4pSAb3. Harmonic control of vibration and sound radiation of a plate using a virtual impedance approach. Nicolas Quaegebeur, Philippe Micheau, and Alain Berry (GAUS, Dept. de Gnie, Univ. de Sherbrooke, 2500 Blvd. de l'Universit, Sherbrooke, QC, Canada J1K 2R1

The present work focuses on the harmonic control of plate vibration and radiation using distributed collocated piezoceramic units. The objective is to add virtual impedance units on the plate and to optimize its complex value for each frequency. In active damping (velocity feedback controllers), the added virtual impedance is equivalent to a resistive effect. In the present study, the added virtual impedance is chosen arbitrarily in the complex plane. Each added impedance unit is composed of a collocated PZT actuator and a PVDF sensor. This approach allows obtaining dual variables for harmonic control. In order to tune the PZT voltage at each frequency, the virtual impedance approach is implemented with a harmonic adaptive control. The stability is always ensured with a centralized processing, and for certain frequencies, a decentralized processing can be used. Preliminary theoretical study is carried out to assess the behavior of the smart panel for different added virtual impedances for a mechanical excitation by a shaker. Those numerical results are compared to experimental measurements performed in an anechoic chamber. Our proposed strategy compare favorably to common strategy of active damping to reduce harmonic sound radiation. Hence, a reduction of 20 dB can be achieved at certain frequencies.

4:00

4pSAb4. Frequency spectra of bilaminar prolate spheroidal shells. Sabih I. Hayek (Dept. of Eng. Sci. and Mech., Penn State Univ., Univ. Park, PA 16802) and Jeffrey E. Boisvert (NAVSEA Newport, Newport, RI 02841)

The kinetic and strain energy densities were derived for the vibration of a bilaminar prolate spheroidal shell of constant thickness. The bilaminar shell is composed of two bonded concentric prolate spheroidal layers of different material properties and thicknesses. The elastic strain energy density has seven independent kinematic variables: three displacements, two thickness shears, and two thickness stretches. Continuity of displacements is enforced at the interfacial reference surface. The shell has constant thickness $h=h1+h2$, where $h1$ and $h2$ denote the thicknesses of the respective layers. The reference surface eccentricity is defined by $1/a$, where a is its shape parameter. Using appropriate comparison functions in terms of Legendre polynomials that satisfy the boundary conditions for a closed spheroidal shell, the system is solved using the Galerkin method. Numerical results of the frequency spectra are presented for various ratios of $h1/h2$ and various material properties for the two layers. Initial results are presented for $a=100$ (a nearly spherical shape). [Work supported by the ONR/ASEE Summer Faculty Research Program.]

4:15

4pSAb5. Noncontact surface wave testing of pavements using microphones. Nils Ryden (Faculty of Eng., Eng. Geology, Lund Univ., Box 118, SE-22100 Lund, Sweden, nils.ryden@tg.lth.se), Michael J. S. Lowe, and Peter Cawley (Imperial College, London SW7 2AZ, UK)

Pavements are constructed using several layers of materials, and their durability depends mainly on the stiffness modulus and thickness of these strata. Surface wave testing is an effective tool to measure the stiffness and thickness of pavement layers. However, measurements are still based on spot testing with fixed receivers and source. The typical large size of pavements and the cost of closing down roads to make stationary testing makes these measurements impractical. We present experiments where a multichannel array of microphones and an automatic source are attached on a small trolley so that measurements can be taken almost continuously while moving. Measurements on asphalt or concrete pavement layers are based on supersonic leaky air-coupled surface waves. We also demonstrate that the same approach can be applied to softer granular pavement layers utilizing the seismic-to-acoustic coupling in poroelastic materials. Results show that microphones can be successfully used to measure correct surface wave dispersion curves even while moving along the surface. This opens up the possibility for faster on-the-fly surface wave testing of pavement layers since surface contact is no longer required. The theoretical background along with experimental results of the application to nondestructive testing of pavements will be presented.

4:30

4pSAb6. Determining source location from vibration "slowness" vectors. Byung-ghun Kim, Jangbom Chai (Dept. of Mech. Eng., Univ. of Ajou, Suwon 443749, Korea, z3hif@naver.com), and Richard H. Lyon (RH Lyon Corp., Belmont, MA 02478-2021)

The time for a wave to transit from one sensor to another is related to its "slowness" and is measured by the phase delay of the cross spectrum of the signals from the sensors. If the sensor separation is much less than a wavelength, this cross spectrum is related to the energy flow in the wave. An "L-shaped" array of three sensors allows one to determine the two vector components of slowness and the direction of the wave. If wave directions are computed at two or more such L-shaped arrays, then it may be possible to "triangulate" and fix a source location. It is possible of course that the directions may be such that there is no solution for a source location. Generally, the wave directions at the arrays are determined for a fixed set of frequencies. Then, at each frequency, a source solution is sought. At some frequencies, there may be no solution so the set of frequencies is diminished. Also, it is possible, even likely, that waves at different frequencies or wavelengths may have taken different paths from the source to the arrays. This paper reviews the results of applying the slowness vector computations to data taken on a model nuclear plant flow loop.

4:45

4pSAb7. Diffuse field interferometry for experimental Green's function estimation and damage detection. Adelaide Duroux, Karim Sabra (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW., Atlanta, GA 30332-0405), James Ayers, and Massimo Ruzzene (Georgia Inst. of Technol., Atlanta, GA)

Structural health monitoring systems often rely on propagating elastic waves through complex structures, which can result in the formation of diffuse-fields over a long reverberation time. Recent theoretical and experimental studies have demonstrated that Green's functions between a pair of monitored points can be extracted from cross-correlation of the recorded diffuse-fields (e.g., scattered fields or ambient noise). Knowledge of the Green's functions between large numbers of points can be used to successfully identify and localize damage in complex structural components. In this work, Green's functions are first estimated experimentally from full-field measurements obtained with a scanning laser vibrometer. This provides the wealth of *a priori* information necessary to detect and localize "secondary" sources, such as damages, when only a limited number of sensors are actually mounted on the structure. The proposed approach relies on the detailed knowledge of the structural response, which is exclusively obtained through experimental measurements performed on the actual component under consideration.

5:00

4pSAb8. Flaw localization in a structure using model-based backpropagation. David Chambers, Sean Lehman, Lisle Hagler, Henry Hsieh, and Karl Fisher (Lawrence Livermore Natl. Lab., P.O. Box 808, L-154 Livermore, CA 94551, chambers2@llnl.gov)

Damage localization is an important part of structural health monitoring. In this talk we present the results of a method of localizing pointlike damage in a structure using changes in the vibrational response induced by the damage. The vibrational response is measured at discrete locations on a structure both before and after damage is induced. The difference in response is used to drive a numerical model of the undamaged structure (backpropagation). The damage location is marked by a peak in the calculated displacement field. Results are shown using laboratory measurements of a cylinder and two nested spherical half-shells. Numerical simulation is used to demonstrate performance in more complex structures. [Work supported by the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract DE-AC52-07NA27344.]

5:15

4pSAb9. Singing sands, musical grains, and booming sand dunes. Tom Patitsas (Dept. of Phys., Laurentian Univ., 935 Ramsey Lake Rd., Sudbury, ON P3E 2C6, Canada)

The origin of the seismic and acoustic emissions from a bed of singing sands or musical grains, impacted by a pestle, is sought in a boundary layer, several millimeters thick, at the leading front end of the pestle. It is assumed that such a layer is the result of the fluidization of the grain asperities due to the high stress level at the pestle front end. Such a fluidization results in a very low modulus of rigidity and in a shear phase velocity about 1 m/s in the boundary layer. The frequencies of the shear modes of vibration, in such a layer, are compared with those determined experimentally. The frequencies depend weakly on the geometry of the leading pestle front. The same concept of the boundary layer can account for the emissions from plates of sand sliding on a dune surface and from grains shaken in a glass jar.

Session 4pSC

Speech Communication: Articulatory Measures and Signal Processing (Poster Session)

Stefan Frisch, Chair
Univ. of South Florida, Communication Sci. and Disorders, 4202 E. Fowler Ave., Tampa, FL 33620-8100

Contributed Papers

All posters will be on display from 1:30 p.m. to 4:30 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:30 p.m. to 3:00 p.m. and contributors of even-numbers papers will be at their posters from 3:00 p.m. to 4:30 p.m.

4pSC1. Speech enhancement for noise-robust speech recognition. Vikramjit Mitra and Carol Espy-Wilson (Inst. for Systems Res., Dept. of Elec. Comput. Eng., Univ. of Maryland, College Park, MD 20742, vmitra@umd.edu)

A modified phase opponency (MPO) model was combined with the aperiodic/periodic/pitch detector (APP) by Deshmukh *et al.* [JASA, **121**, 3886–3898 (2006)] for speech enhancement. The MPO-APP passes the speech-dominated regions as is but greatly attenuates the noise-dominated regions. Thus, the noise in nonspeech regions and in the regions between the formants are significantly reduced. Unlike many other techniques, the MPO-APP does not introduce musical noise. However, given that the noise in formant regions is passed along with the speech signal, the enhanced signal sometimes has a shadow effect. To get rid of this shadow effect, we recently developed a preprocessor for the MPO-APP. First, the signal to noise ratio of the signal is estimated. Second, we use an adaptive generalized spectral subtraction technique to reduce the noise in the speech dominant regions. The results show a significant improvement in the quality of the enhanced signal and, in addition, increased accuracy in automatic recognition of the enhanced signals over that obtained from the original signals or the signals enhanced without the preprocessor. [Research supported by NSF Grant IIS0703859.]

4pSC2. Variable resolution spectral/temporal features for automatic speech recognition. Stephen Zahorian and James Wu (Dept. of Elec. and Comput. Eng., Binghamton Univ., Binghamton, NY 13903, zahorian@binghamton.edu)

Many studies from speech science have shown that the mel frequency scale more closely matches speech perception than the linear frequency scale. Automatic speech recognition engineers have empirically demonstrated that the use of the mel scale results in more accurate speech recognition than that obtainable with features computed with respect to a linear frequency scale. The features most typically used for automatic speech recognition are mel frequency cepstral coefficients (MFCCs), along with delta and acceleration terms that represent the temporal evolution of MFCC over very short time intervals. However, the MFCC features do not encode the better temporal resolution that is possible at higher frequencies with low-frequency resolution. In this paper, a two-dimensional feature set is presented that incorporates good frequency resolution and low-time resolution at low-frequencies, and low frequency resolution and good time resolution at high frequencies. These features are computed from overlapping blocks with an effective length of approximately 100 ms at low frequencies and approximately 20 ms at high frequencies. Experimental results are given for phonetic recognition using the TIMIT database. The implications are that features that encode the temporal evolution of speech spectra are important for automatic speech recognition.

4pSC3. Modeling prosodic rhythm: Evidence from second language speech. Emily Nava, Louis Goldstein (Linguist. Dept., Univ. of Southern California, GFS 301, Los Angeles, CA 90089, nava@haskins.yale.edu), Hosung Nam, Michael Proctor, and Elliot Saltzman (Haskins Labs., New Haven, CT 06511)

The global prosodic structure of languages has been described using the typological dichotomy of stress-timed versus syllable-timed. Various indices have been successfully employed in literature for quantifying these classifications, one of which is the duration ratio between the total voiceless and total voiced stretches in the signal. It has been further shown that various language-specific characteristics, such as syllable-structure phonotactics and stress-sensitive lengthening and shortening, can contribute to this difference. To reveal the interaction of these components, acoustic data from running speech of L1 Spanish/L2 English and native English speakers were analyzed. Total ratio of voiceless-to-voiced durations discriminated L1 Spanish (lower) and L1 English (higher); L2 speakers showed ratios in between the two, with higher proficiency L2 speakers showing ratios closer to L1 English. A task dynamic application, a speech planning and production model [Nam et al., J. Acoust. Soc. Am. **115**, 2430 (2004)] was used to model the performance of the L1 and L2 speech of L2 speakers, allowing determination of which contributors to global rhythm are more readily acquired by L2 speakers. [Work supported by NIH and NSF.]

4pSC4. Comparison of a new frequency domain periodicity detector derived from a temporally stable power spectral representation and zero frequency filtering based periodicity detector. Hideki Kawahara, Hanae Itagaki, Ryuichi Nisimura, and Toshio Irino (Faculty of Systems Eng., Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan, kawahara@sys.wakayama-u.ac.jp)

Recently, an event based periodicity detection method was proposed using zero frequency filtering method [Yegnanarayana *et al.*, ITRW Aalborg 2008]. The authors also introduced a new periodicity detector based on TANDEM-STRAIGHT [Kawahara *et al.*, ICASSP2008 (2008)], a combination of temporally stable power spectrum estimation method for periodic signals and a spectral envelope recovery method based on consistent sampling [Unser, Proc. IEEE (2000)]. These methods provide a complementary set of information on excitation source signals of speech sounds. MATLAB implementation of these methods was evaluated using publicly available EGG databases and compared with popular existing methods. It was found that the zero frequency filtering based method runs eight times faster than real time even with MATLAB implementation and yielded comparable performance to popular methods. The TANDEM-STRAIGHT based methods also yielded comparable performance and additional information that is useful to represent diplophonia, for example. Detailed analysis examples and their application for TANDEM-STRAIGHT will also be discussed.

4pSC5. Vowel and speaker classification based on multivowel linear discriminant classfication. Al Yonovitz (Dept. of Communicative Sci. and Disord., Univ. of Montana, Missoula, MT 59812, al.yonovitz@umontana.edu), Ryan Anderson (Univ. of Montana, Missoula, MT 59812), and Sarah Van Orden (Univ. of Montana, Missoula, MT 59812)

Accurate and automated voice or speaker identification has been a major goal for those involved in forensic issues. In addition, voice and speaker identification has many applications in security and business. Numerous previous efforts to derive features for speaker classification have failed to achieve a sufficiently low-error rate. In this study a linear discriminant analysis was independently performed for vowel formants ($F1$, $F2$, $F3$) for

each of ten vowels. The standardized canonical discriminant coefficients (SCDCs) weight each of the formants toward the group separation. These SCDC values are then linearly combined to form a single scalar for each vowel. An analysis that considered the multivariate data vector composed of the ten vowel scalars was then used for classification. All analysis was accomplished with the classic archived data set of Peterson and Barney (1952) and an additional data set derived for this study. The classification was over 98% accurate in separating males (M), females (F), and children (C) into subgroups, greater than any other study has provided. The methodology was used in a manner which tested specific speaker identification.

4pSC6. Reconstruction of missing formants based on spectral power series expansion. Kesaaki Minemura (Dept. of Comput. Sci., Waseda Univ., 3-4-1 Okubo, Shinjuku-ku, Tokyo 169-8555, Japan, minemura@shirai.cs.waseda.ac.jp), Satoru Goto, and Mikio Tohyama (Waseda Univ., Saitama 367-0035, Japan)

In this paper, we investigated spectrum estimation for vowels using power series expansion. In general, sound signals can not keep their characteristics when the spectral peaks and dips are missing. Consequently, the missing spectral peaks and dips are estimated by using power series expansion. First, vowels are divided into cycle by cycle. Second, the one-cycle vowels are converted into frequency by FFT. Third, damaged spectra were made for demonstrating spectrum estimation. Fourth, we estimated missing spectral peaks and dips using power series expansion. In our spectrum estimation, the differential coefficients for power series expansion were calculated from nearby spectrum of missing spectrum. In this experiment, the estimated spectrum was almost the same compared with the original spectrum in the case of approximation by power series expansion on the rank of 6. The spectrum is restored about the range of 180 Hz using nearby 10-Hz samples. Consequently, power series expansion can estimate spectral peaks and dips for recovering spectrum.

4pSC7. Multipitch tracking and speaker separation for two or more speakers. Srikanth Vishnubhotla and Carol Espy-Wilson (Dept. of Elec. Eng., Inst. for Systems Res., A V Williams Bldng., Univ. of Maryland, College Park, MD 20742, srikanth@umd.edu)

Accurate robust pitch tracking in multispeaker environments is an important issue in monaural speech separation. This research presents a spectrotemporal multipitch algorithm designed to detect voiced unvoiced regions in a mixture of multiple speakers, to identify the number of speakers in voiced regions, and to yield the pitch estimates of voiced speakers. Pitch estimation is based on identifying the minima of the multidimensional (MD) AMDF. The number of voiced speakers is determined by the temporal evolution of the MD AMDF. The algorithm has been developed and tested for the two-speaker case and is being extended to the three-speaker case. Evaluation on a framewise basis for the two-speaker case yielded an insertion error rate of 1.78%, a deletion error rate of 14.37%, and a substitution error rate of 3.26%, comparable to the state of the art. Pitch estimation errors resulted primarily from speaker domination. The extension to the three-speaker case shows promising preliminary results. The proposed algorithm can contribute to speaker separation by helping to identify spectrotemporal regions dominated by one of the speakers from those that have comparable contribution from all speakers. Appropriate strategies can then be used to separate speech in these different scenarios. [Research supported by NSF Grant BCS-0519256].

4pSC8. Experimental and numerical determination of the surface deformation of a synthetic model of the human vocal folds. Li-Jen Chen and Luc Mongeau (Dept. of Mech. Eng., McGill Univ., McDonald Eng. Bldg., 817 Sherbrooke St. West, Montreal, QC H3A 2K6, Canada, ljchen @purdue.edu)

A model of human vocal folds was fabricated using a three-component liquid platinum-catalyzed silicone solution. The size, idealized shape, and mechanical properties of the homogeneous synthetic model were selected based on the available data. The superior surface displacement of the synthetic model during self-oscillations was measured using the digital image correlation technique. A finite element model of the synthetic model was created to calculate the state of the deformable solid. Modal testing of the synthetic model was performed to establish the material properties and to verify boundary conditions in the simulation. The self-oscillation of the synthetic model was simulated by imposing a sinusoidal pressure loading over model surfaces, with frequency and amplitude determined from the direct measurement. From the simulation, the von Mises stress over the inferior surface was found to be around 2.2 kPa during the maximum orifice opening, which is around twice of that over the superior surface. So far, only the superior surface deformation data have been available because of technical limitations in clinical studies. The current study may provide additional information, such as the maximum amplitude and location of the peak stress, which is useful for diagnostic and treatment purposes. [Work supported by NIH.]

4pSC9. Recent improvements to the University of California, Los Angeles' voice synthesizer. Norma Antonanzas-Barroso, Jody Kreiman, and Bruce R. Gerratt (Div. Head/Neck Surgery, UCLA School of Medicine, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794, nab@ucla.edu)

A number of enhancements have recently been added to the UCLA voice synthesizer. Additions have been made to add functionality and to address several theoretical issues that arose during development and application. New functions include the ability to manipulate the source spectrum directly by changing the slope and/or amplitude of a user-defined group of harmonics or by manipulating individual harmonics directly. The synthesizer also allows users to add zeros to the vocal tract transfer function, improving the modeling of many pathological voices. A number of other enhancements improve the ability to create and systematically vary stimuli for perceptual experiments. Theoretical development has focused on the importance of pitch-synchronous Fourier analysis in modeling the voice source, particularly with respect to measuring and manipulating the amplitudes of $H1$ and $H2$. Issues surrounding spectral effects of upsampling, downsampling, and pulse stretching, which are needed for precise manipulation of $F0$, will also be discussed. This synthesizer is available with documentation as open source shareware at www.surgery.medsch.ucla.edu/glottalaffairs/, and copies will be available at the conference. [Work supported by NIH/NIDCD Grant No. DC01797.]

4pSC10. A survey of respiratory system behavior during pauses in spontaneous speech. Janet Slifka (MGH Voice Ctr., 11th Fl., One Bowdoin Square, Boston, MA 02114)

This work is part of an ongoing study to characterize respiratory system involvement during the generation of pauses in connected speech. At pauses, the speaker takes a breath or the speaker does not take a breath. Previously reported results for read speech observed that without-breath pauses were generated with a sharp movement toward net inspiratory effort followed by a sharp return toward increased net expiratory effort [J. Slifka, *In Dynamics of Speech Production and Perception* (IOS, 2006), pp. 45-58]. In the present work, 52 utterances of spontaneous speech from four speakers were analyzed. Behaviors in spontaneous speech include not only the activity similar to that observed for read utterances but also additional types of behaviors. Some without-breath pauses were observed to have regions of oscillation in net effort. Such behavior appears to be more common during longer pauses and during hesitations. Secondly, some without-breath pauses were generated with a reversed pattern—sharp movement toward increased net expiratory effort and back to lesser net expiratory effort. This pattern was much less common and appears to be associated with relatively short pauses in regions of hesitations. These results characterize the respiratory system as an active component in speech production. [Work supported by NIH-NIDCD DC007986]

4pSC11. Modulation of middle and long-latency cortical potentials during pitch-shifted auditory feedback perturbation. Roozbeh Behroozmand, Hanjun Liu, Laura Karvelis, and Charles R. Larson (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208)

It has been shown that applying perturbation to the real-time voice auditory feedback evokes compensatory responses that minimize the vocal output error. Auditory feedback-based control of the voice fundamental frequency involves the integration of multiple neural substrates that take part in the process of vocal output error detection and correction. Previous studies have shown that the cortical brain potentials are modulated during active vocalization versus passive listening to the self-generated voice in the absence of auditory feedback perturbation. To learn more about these central mechanisms, the present study tested the effect of vocalization condition across different pitch-shifted auditory feedback magnitudes. Twenty English speaking young adults vocalized a vowel sound and received pitch-shifted feedback of +100, 200, or 500 cents while scalp potentials were recorded. Results show that the middle-latency P50 responses were significantly larger during the active vocalization task only at 200 cents frequency shifts. Long-latency N100 and P200 peaks were significantly larger during vocalization for 100 cents and 100 cents/200 cents, respectively. The overall conclusion is that the middle- and long-latency cortical responses are significantly greater during vocalization, but the difference between response peak magnitudes across vocalization and passive listening conditions diminishes with larger pitch frequency shifts in the voice auditory feedback.

4pSC12. Hemisphere lateralization underlying auditory feedback control of voice *F*0. Hanjun Liu, Michelle Meshman, and Charles Larson (Dept. Commun Sci. Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL, 60208)

Reflexive compensation for unexpected pitch changes in auditory feedback for stabilization of voice fundamental frequency ($F0$) has been demonstrated using the frequency perturbation technique. However, neural mechanisms of the pitch-shift reflex are still unclear. This study investigated hemisphere lateralization in the auditory processing of pitch changes by using event-related potentials. During vocalization of sustained vowels, subject's voice pitch feedback was randomly shifted upward or downward 100, 200, or 500 cents with 200 ms duration. Simultaneously, brain activity was recorded through surface electrodes located on the scalp bilaterally referenced to linked earlobes. The results showed larger brain potentials (peak difference between $P200$ and $N100$) in the $C3$ (left) than in the $C4$ (right) site, indicating left hemisphere dominance in the central motor area, while right hemisphere dominance was found in the lateral frontal area with larger potentials in the $F8$ (right) than in the $F7$ (left) site. Furthermore, these patterns were maintained regardless of stimulus magnitude and stimulus direction. This preliminary study suggests that, although both left and right hemispheres are involved in voice production, functional lateralization of the brain may exist for voice $F0$ control. Results also suggest that lateralization varies along the anterior-posterior dimension of the neural axis.

4pSC13. Measuring glottal area and flow in an excised canine larynx model using stereoscopic particle imaging velocimetry. Ephraim Gutmark (Univ. of Cincinnati, 799 Rhodes Hall, P.O. Box 210070, Cincinnati, OH 45221, ephraim.gutmark@uc.edu) and Shanmugam Murugappan (Univ. of Cincinnati Medical Ctr., Cincinnati, OH 45267-0528)

As opposed to the glottal area waveform, the opening and closing phases of glottal flow are asymmetric. Near the end of the closing phase, flow exiting the glottis has a rapid deceleration not seen during the opening phase; this "skewing" of the flow waveform is important for producing loudness and higher frequency harmonics. The major mechanism for the skewing of the flow waveform, relative to the area waveform, has been attributed to the nonlinear interaction between the vocal tract and the glottal flow. Previous work from this laboratory supports a hypothesis that both area and flow skewing are seen even without the vocal tract due to intraglottal flow separation vortices producing relative negative pressure. To help test this hypothesis in an animal model, a technique is necessary that can reliably measure glottal area and flow. In this work, we will show results for both glottal area and flow during phonation, in excised canine larynges, using stereoscopic particle imaging velocimetry and high speed visualization. The results support the hypothesis that intraglottal flow separation can be one mechanism for producing both flow and glottal skewing. [Work supported by NIDCD 5K08DC005421.]

4pSC14. Unsteady laryngeal airflow simulations: An analysis of the generated intraglottal vortical structures. Mihai Mihaescu (Dept. of Aerosp. Eng. and Eng. Mech., Univ. of Cincinnati, 310 Rhodes Hall, ML 0070, Cincinnati, OH 45221, Mihai.Mihaescu@uc.edu), Sid Khosla, and Ephraim Gutmark (Univ. of Cincinnati, Cincinnati, OH 45221)

Unsteady flow simulations in diffuserlike vocal-fold models proved that laryngeal airflow generates vortical structures in the intraglottal region. However, the features of these vortices and their influence on the vocal-fold motion were not clearly addressed. The present work characterizes the intraglottal vortical structures developed during the closing phase of the phonation cycle in order to analyze their possible influence on the voice quality. It is shown that intraglottal vortices are formed on the divergent slope of the glottis, just downstream of the separation point. The core of the vortical structures is characterized by important negative static pressure values (lower as compared with the surrounding pressure field). These vortices gain strength and increase in size as are convected downstream by the flow. The mechanism for which the intraglottal flow structures are becoming stronger is attributed to the entrained air from the supraglottal region. The instantaneous pressure loads on the divergent glottal slope are not uniform (in both time and space) and dependent on the vortical structures traveling nearby the glottis. The negative static pressures associated with the intraglottal vortical structures suggest that the closing phase during phonation may be accelerated by such vortices.

4pSC15. Articulatory timing in sentence production in young normal speakers and speakers with apraxia of speech: A speed history analysis. Kana Taguchi, Michiko Hashi (Dept. of Commun. Disord., Prefectural Univ. of Hiroshima, 1-1 Gakuen-cho Mihara, Hiroshima 723-0053, Japan p724015ue@pu-hiroshima.ac.jp), and Katharine Odell (Marshfield Clinic, Marshfield, WI 54449)

Apraxia of speech (AOS) is typically considered as characterized by kinematic timing problems among articulators; however, kinematic descriptions of such phenomenon are scarce. The study describes kinematic timing based on a speed history of the lower lip and the jaw, across speaking rates in young normal speakers and speakers with AOS. Kinematic data were acquired through the x-ray microbeam system, and data of young normal speakers were taken from the x-ray microbeam speech production database. A short sentence was the speech material/target. Acoustic correlates of major speed peaks of the jaw and the lower lip, as well as their temporal relationships with each other, were investigated. Of particular interest was how changes in speech rate affect such temporal relationships in normal speakers and speakers with AOS. The goal of this study was to establish a description method of inter- and intra-articulator timing for use in analyses of point-parametrized articulatory data in sentence-level materials.

4pSC16. Experimental study and theoretical simulation of stress relaxation behavior of vocal fold lamina propria tissue. Yu Zhang, Megan Keuler, and Jack Jiang (Dept. of Surgery, Div. of Otolaryngol. Head and Neck Surgery, Univ. of Wisconsin Med. School, Madison, WI 53792-7375)

The viscoelastic properties of vocal fold lamina propria tissue play an important role in tissue modeling, as well as in clinical studies of the effects of carcinoma, scarring, atrophy, or edema on dysphonia. Although the impact of these properties has been well studied, quantification of viscoelastic behaviors such as creep and stress-relaxation continue to be important in the research. The current study examined the stress-relaxation curves of ten canine vocal fold cover samples by stretching the tissue to 5%, 10%, 15%, and 20% of the sample reference length. The force on the lamina propria tissue was then recorded for 5 mins, and graphed so the stress-relaxation response could be seen. The curves obtained were what we hypothesized, with the lamina propria tissue exhibiting exponential relaxation initially, and very slow decay at the end of the trial. These results are consistent with our previous studies on stress-relaxation predicted by the biphasic theory of vocal fold tissue [Zhang *et al.*, J. Acoust. Soc. Am. **123**, 1627–1636 (2008)]. Finite element simulation has also been given to investigate the stress distributions within the vocal fold epithelium and lamina propria tissues.

4pSC17. Videokymographic analysis of irregular vocal fold vibration in laryngeal paralysis. Miwako Kimura, Niro Tayama (Otolaryngol. and Tracheo-esophagology, Int. Medical Ctr. of Japan, Tokyo, Japan and Univ. of Tokyo, Tokyo, Japan), and Roger W. Chan (Univ. of Texas Southwestern Medical Ctr., Dallas, TX)

Despite recent advances in high-speed digital imaging methods, laryngeal stroboscopic imaging is still commonly used for the clinical diagnosis and assessment of voice disorders. Yet standard videostroboscopy, designed for the examination of periodic or near-periodic vibration, is unable to provide detailed information on irregular vocal fold vibratory patterns. This study examined the vocal fold vibratory patterns of two patients with unilateral vocal fold paralysis with high-speed digital kymographic imaging. Laryngeal vibration was examined before and after the medialization procedure of arytenoid adduction, and relevant features were extracted from the videokymographic images. Results demonstrated two distinct vibratory frequencies for the contralateral vocal folds in both subjects prior to the procedure (269 Hz vs 361 Hz, and 114 Hz vs 154 Hz), but a single vibratory frequency following medialization (361, and 154 Hz). The restoration of vibratory symmetry after arytenoid adduction indicated that this procedure could significantly improve the synchronization and entrainment between asymmetric vocal folds, in parallel with improvements in glottal competence.

4pSC18. Perceptual cues for consonant identification. Feipeng Li and Jont Allen (Beckman Inst., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave, Urbana, IL 61801)

This research quantitatively explores the perceptual cues of initial consonants by using psychoacoustical methods. Speech sounds are encoded by across-frequency temporal onsets called events. To determine the time-frequency importance function of the consonant sounds, speech stimuli (16 nonsense CVs from the LDC-2005S22 database) are high-pass or low-pass filtered and time-truncated before being presented to normal hearing listeners. Databases of speech perception under various signal to noise ratio (SNR) conditions are constructed to investigate the effect of noise on speech recognition. A visualization tool that simulates the auditory peripheral processing, called the AI-gram, is used for the analysis of the speech events under various SNR conditions. To verify the nature of the events, a special software has been developed to convert one sound into another, starting from real speech sounds, by removing primary noise robust cues. In pilot studies with a hearing-impaired subject, it is shown that feature boosting improves the robustness of select speech consonants to noise.

4pSC19. The effect of articulatory placement on acoustic characteristics of nasalization. Panying Rong and David P. Kuehn (Dept. of Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S. Sixth St., Champaign, IL 61820, prong2@uiuc.edu)

Vowels adjacent to nasal consonants (/m/, /n/, /ŋ/ in English) tend to be nasalized. The effect of velopharyngeal opening on vowel nasalization has been studied by Feng *et al.* (1996) and Pruthi *et al.* (2007) with a transmission line model of the vocal tract based on MRI data of the area function. In order to find out the effect of articulatory placement on acoustic properties of nasalized vowels, the current study simulated the transfer function of coarticulated vowels in different nasal-vowel utterances. The results revealed different acoustic characteristics of the same vowel in various nasal contexts, which demonstrated the effect of articulatory placement on vowel nasalization and suggested the possibility for oral articulation to compensate for spectral changes caused by failure of velopharyngeal closure among hypernasal patients. The articulatory parameters for compensatory articulation were optimized by minimizing the spectral differences (first four formants) between the compensated nasalized vowel and its oral counterpart. The nasalized vowel was then synthesized with an articulatory synthesizer and evaluated for vowel quality and nasality in the following perceptual study.

4pSC20. The effects of coarticulatory timing and lexical frequency on vowel nasalization in English: An aerodynamic study. Jason B. Bishop (Dept. of Linguist., UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, j.bishop@ucla.edu)

Recently it has been suggested that a certain degree of variability in coarticulatory vowel nasalization is due to variation in the temporal alignment of nasal and oral gestures for N. In particular, the extent of vowel nasalization in VNC sequences is shown to be inversely related to the duration of the oral gesture for the nasal. Beddor [nasals and nasalization: the relation between segmental and coarticulatory timing, ICPhS (2007) present acoustic data which suggests this to be the case in English; in environments where nasals are shorter, such as VNC[−voice] versus VNC[+voice], anticipatory nasalization is longer. The present study examines vowel nasalization in such environments and attempts to corroborate the aforementioned acoustic findings with aerodynamic data. Additionally, other possible sources of variation in the extent of anticipatory nasalization are explored, namely, the effects of lexical frequency, which has been claimed to correlate negatively rather than positively with coarticulation [carborough (2004).]

4pSC21. Children form tense and lax vowel classes under aerodynamic constraints. Piers Messum (112 Warner Rd., London SE5 9HQ, UK, p.messum@gmail.com)

In English, lax vowels contrast with tense vowels as follows: (1) They are always "checked" by a following consonant, (2) they require only moderate displacement of the tongue from its resting position, and (3) they are "short." If these characteristics are arbitrary and independent–as currently believed–then it is a remarkable coincidence that each divides the vowel inventory into classes with the same membership. Alternatively, these characteristics emerge together under constraints in child speech that do not appear in the adult model. Aerodynamically, the pressures in child speech are higher, flows are similar, and airways are smaller. Mechanically, the respiratory drive that supports speech is pulsatile rather than smooth. This latter difference is heightened in stress-accent languages, such as English, where a child must reinforce pulses for greater loudness on stressed syllables. The constraints created by these factors require a child to check vowels made with open articulations and lengthen those made with close ones. The former behavior protects the subglottal pressure head. The latter is the indirect result of limiting airflow to avoid unwanted turbulent noise at the point of maximum oral constriction. Increasing laryngeal resistance to do this prolongs the time it takes to dissipate a pulse.

4pSC22. Interarticulator timing in a sentence production in young normal speakers and speakers with apraxia of speech. Mayuko Fujishita, Michiko Hashi (Dept. of Commun. Disord., Prefectural Univ. of Hiroshima, 1-1 Gakuen-cho, Mihara, Hiroshima 7230053 Japan, p724019fa@pu-hiroshima.ac.jp), and Katharine Odell (Marshfield Clinic, Marshfield, WI 54449)

Apraxia of speech (AOS) is typically considered as characterized by kinematic timing problems among articulators; however, kinematic descriptions of such phenomenon are scarce. The goal of this study was to describe articulatory timing between the lips and jaw and the lips and tongue in young normal and AOS speakers across three speaking rates, using x-ray microbeam data. Data of young normal speakers were taken from the x-ray microbeam speech production database. Lip protrusion and tongue elevation timings were derived from the kinematic and acoustic data in the production of "too" in the sentence "The other one is too big" and lip and jaw closing timings for /b/ and /w/ were derived from the kinematic data of the same sentence. The temporal relationships between these articulatory movements in each event were then examined with consideration of the effect of speaking rate. The results are discussed in relation to the effect of speaking rate on interarticulator timing and interspeaker variability in interarticulator timing.

4pSC23. Articulatory and acoustic measures of vowel frontness in a study of velar-vowel coarticulation. Sylvie M. Wodzinski, Gena Rizzitano, and Stefan A. Frisch (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave. PCD1017, Tampa, FL 33620, wodzinsk@mail.usf.edu)

Previous work has found a strong correlation between the frontness of closure location for velar stops (measured manually from ultrasound images by a trained expert) and the frontness of the following vowel (measured by F2). In this study, semiautomatic measures of tongue frontness for vowels were made from ultrasound images of the vowel articulation. However, it was found that the acoustic measure, F2, correlated more closely with the frontness of the preceding consonant than any of the ultrasound based articulatory measures of the vowel. Explanations for why an acoustic measure of coarticulation would be better than an articulatory measure will be discussed. It may be that the ultrasound based measures do not adequately capture retraction of the tongue root, which would influence F2 and presumably affect coarticulation with the consonant. Surprisingly, it was also found that the frontness measures for the consonant were more highly correlated with F2 than the frontness measures for the vowel, suggesting that the tongue frontness measures may be indirectly affected by tongue height in some way. Overall, the manual measures of an expert appear to be superior to semiautomatically generated measures.

4pSC24. Semiautomatic measures of velar stop closure location using the EdgeTrak software. Sabrina J. McCormick, Stefan A. Frisch, and Sylvie M. Wodzinski (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave., PCD1017, Tampa, FL 33620, smccorm3@mail.usf.edu)

Previous work has found a strong correlation between the frontness of closure location for velar stops (measured manually from ultrasound images) and the frontness of the following vowel (measured by F2). In this study, semi automatic measures of tongue frontness during a velar closure were made. Tongue edge traces were made using the EDGETRAK software (Li *et al.*, 2005, Clinical Linguistics and Phonetics, 545–554). Frontness was then quantified from these traces using three different measures: Bressman's anteriority index (Bressman *et al.*, 2005, Clinical Linguistics and Phonetics, 573–588), a modified version of the anteriority index created for this study, and a measure of the center of mass of the tongue created for this study. When compared to the original manual measures, the modified anteriority index correlated most highly with both the manual measurement of the consonant closure location and also with F2 of the following vowel. The modified anteriority index uses an angle based weight in the anteriority calculation (as opposed to the arbitrary weights of Bressman's anteriority index). The center of mass was the worst performing measure, and it appeared to be overly sensitive to the extreme anterior and posterior portions of the tongue edge trace.

4pSC25. Tongue body movements in speech: Straight or curved paths? Anders Lfqvist (Haskins Labs., 300 George St., New Haven, CT 06511 and Dept. Logopedics, Phoniatrics, Audiol., Lund Univ., Lund, Sweden, lofquist@haskins.yale.edu)

This paper examines tongue body movements between two vowels with particular emphasis on the shape of the movement paths. Earlier work on tongue movements in speech has mostly focused on movements for consonants. The movements analyzed are from the first to the second vowel in a sequence of vowel-bilabial consonant-vowel. Native speakers of Japanese and Italian served as subjects. The linguistic material consisted of words with a long or short labial consonant. Recordings were made using a magnetometer. To assess the movement path, the movement magnitude was calculated in two ways as a straight line, the Euclidean distance, and as the actual path, obtained by summing the individual Euclidean distances between successive samples from movement onset to offset. The ratio between the path and the Euclidean distance is 1 when the movement is a straight line and greater than 1 when the path is curved. For most of the movements, the ratio was less than 1.1, thus suggesting that these movements are almost a straight line. There was no clear difference between movements during long and short labial consonants. [Work supported by NIH.]

4pSC26. Evidence for interaction between speech rhythm and gesture. Sam Tilsen (Dept. of Linguist., Univ. of California, Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, tilsen@berkeley.edu)

Temporal patterns in speech occur on multiple timescales. Speech gestures generally occur on a fast timescale; previous work has found evidence for a *c*-center effect in complex syllable onsets (e.g. /spa/), whereby initiations of tongue blade and lip movements associated with /s/ and /p/ are equally displaced in opposite directions from initiation of tongue body movement associated with the vowel [C. Browman and L. Goldstein, Phonetica **45**, 140–155 (1988)]. Speech rhythms occupy a slower timescale; in metronome-driven phrase repetition tasks, rhythmic timing is more variable for higher-order target ratios of intervals between stressed syllables and phrases [F. Cummins and R. Port, J. Phonetics **26**, 145–171 (1998)]. An experiment was conducted to investigate how these gestural and rhythmic patterns interact. Gestural kinematics were recorded using electromagnetic articulometry during a repetition task with the phrase "take on a spa." Significantly greater variance in relative timing of tongue and lower lip movements associated with /s/ and /p/ was observed with the more difficult (less harmonic) target rhythms. Within-gesture effector synergies between jaw and tongue differed systematically across rhythmic conditions. These results demonstrate a substantial interaction between rhythmic and gestural systems. Observed variability patterns are simulated with a dynamical model of phase-coupled oscillators.

THURSDAY AFTERNOON, 13 NOVEMBER 2008 — LEGENDS 12, 2:00 TO 3:45 P.M.

Session 4pSP

Signal Processing in Acoustics: Target Tracking and Beamforming

Charles F. Gaumond, Chair
Naval Research Lab., 4555 Overlook Ave., S.W., Washington, D.C. 20375-5350

Contributed Papers

2:00

4pSP1. Multitarget tracking using acoustic arrays. R. Daniel Costley, Jay E. Williams, Robert C. Clark, Matthew A. Gray, Zachary Williams, Gary Harrington (Miltec Res. and Technol., A Div. of Miltec Corp., A Ducommun Co., 9 Industrial Park Dr., Oxford, MS 38655), William G. Frazier (Independent Consultant, Oxford, MS 38655), and Jere Singleton (U.S. Army Space Missile Defense Command, Huntsville, AL)

A tracking system has been developed, which consists of three or more five-microphone arrays. In previous experiments, these arrays have been deployed and used to track broadband, high-speed (subsonic) airborne, acoustic sources. Recent developments to this system have been made so that multiple acoustic sources can be tracked simultaneously. A field test was conducted this past year, which demonstrated this capability by tracking two low-flying learjets. The tracking system consisted of six acoustic arrays spaced approximately 400 m apart. The bearing angles from each array to the acoustic sources were determined from the blind source separator (BSS) algorithm. The BSS utilizes a mathematical propagation model along with an optimization technique, which minimizes the difference between the model and the measured data to yield a bearing to the target. The bearing angles from two or more arrays are transmitted to a master node, which uses this information to estimate the position, speed, and heading of the targets using a tracking algorithm based on the extended Kalman filter algorithm. The field test will be described and the results will be presented and discussed.

2:15

4pSP2. A novel approach for the localization of sound sources. Piervincenzo Rizzo and Giacomo Bordoni (Dept. of Civil and Environ. Eng., Univ. of Pittsburgh, 949 Benedum Hall, 3700 O'Hara St., Pittsburgh, PA 15261, pir3@pitt.edu)

Targeting people or objects by passive acoustic sensors is of relevant interest in several military and civil applications, spanning from surveillance

and patrolling systems to teleconferencing and human-robot interaction. To date methods and patents focused solely on the use of beamforming algorithms to compute the time of arrival of sounds detected by using omnidirectional microphones that are sparsely deployed. This paper describes the preliminary results of a novel approach devoted to the localization of ground borne acoustic sources. It is demonstrated that an array made of at least three unidirectional microphones can be exploited to detect the position of the source. Pulse features extracted either in the time domain or in the frequency domain are used to identify the direction of the incoming sound. This information is then fed into a semianalytical algorithm devoted to the identification of the source location. The novelty of the method presented here consists of the use of unidirectional microphones rather than of omnidirectional microphones and of the ability to extract the sound direction by considering features such as the pulse amplitude rather than the pulse arrival time. It is believed that this method may pave the road toward a new generation of reduced size sound detectors and localizers.

2:30

4pSP3. Real-time noise source identification using field programmable gate array technology. Kurt Veggeberg (Natl. Instruments, 11500 N. Mopac C, Austin, TX 78759, kurt.veggeberg@ni.com)

Acoustic beamforming is usually used as an off-line analysis tool for noise source identification (NSI). The computational requirements of beamforming makes real-time processing difficult to achieve in conventional NSI measurement systems. This limitation prevents conventional NSI measurement systems for applications such as on-line monitoring. This paper presents the theory, design, and development of a real-time NSI measurement system based on reconfigurable input/output field programmable gate array (FPGA) technology. Some techniques used in applying beamforming analysis methods with an FPGA are proposed. Signal processing includes time filtering of input data and spatial beamforming with special treatment to obtain the best result. Pipelining method is used to make full use of the FPGA resources and reach high performance for real-time applications. Testing examples are given to demonstrate the application for NSI.

2:45

4pSP4. Effects of multiple contacts on cued beamforming in active-passive data fusion. T. W. Yudichak and Bryan A. Yocom (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, yudi@arlut.utexas.edu)

Active-passive data fusion seeks to combine the information measured on multiple sonar sensors to detect and track contacts more effectively than is possible with individual active or passive sensors. One way in which sonar signal processing can benefit from data fusion is cued beamforming, the allocation of beamforming resources based on the current estimate of the location of a contact. Cued adaptive beamforming (ABF) on passive arrays has been shown to provide more precise direction-of-arrival estimates than standard ABF in some circumstances involving a single contact of interest. At the same time, the performance of cued beamforming in directions away from the contact can be severely degraded relative to standard beamforming, resulting in the possible loss of detection of other contacts. This talk investigates the trade-offs between the gain in information on a single contact and the loss of information on other contacts when cued ABF is applied to passive arrays within a multisensor data fusion framework. The effectiveness of cuing different arrays with different contacts as well as alternating between standard and cued beamforming is examined and discussed. [Work supported by ONR.]

3:00—3:15 Break

3:15

4pSP5. Directional sources and beamforming. Christian Bouchard (SITE, Univ. of Ottawa, 800 King Edward, Ottawa, ON K1N 6N5, Canada and Inst. for Microstructural Sci., Natl. Res. Council of Canada, Ottawa, ON K1A 0R6, Canada, christian.bouchard@nrc-cnrc.gc.ca), David I. Havelock (Natl. Res. Council of Canada, Ottawa, ON K1A 0R6, Canada), and Martin Bouchard (Univ. of Ottawa, Ottawa, ON K1N 6N5, Canada)

Beamforming is done with an array of sensors to achieve a directional or spatially specific response. It relies on a model of the wave front (source model) arriving at the array to calculate the time delay, or frequency domain phase shift, that must be applied to the signal of each sensor so that they may be summed coherently. Beamforming may be used to improve signal to noise ratio, reduce reverberation, cancel interference, or estimate source location. In this talk the directionality of some real world sources that deviate from an ideal point source is discussed. Performance measures used to evaluate the directivity properties of a beamformer are reviewed. The validity of assuming a point source is examined and challenges for beamforming with nonpoint sources are discussed.

3:30

4pSP6. Robust direction-of-arrival estimation by understanding global acoustic scene. Mitsunori Mizumachi (Kyushu Inst. of Tech., 1-1 Sensuicho, Tobata-ku, Kitakyushu-shi, Fukuoka 805-8550, Japan, mizumach@ecs.kyutech.ac.jp)

Direction-of-arrival (DOA) is an important clue in acoustic signal processing. It is, however, difficult to accurately estimate DOAs in the presence of acoustic interference such as background noises and reverberation. The author has proposed a robust DOA finder with an environmental noise model, which describes spectral characteristics of background noises. Knowledge on acoustic interferences helps to make DOA estimation robust, but is difficult to be estimated exactly. The method works well only using the dominant subband components, in which the target signal is distinguished compared with interferences, in some noisy conditions. The noise model is updated time by time based on the DOA estimates to cope with nonstationary noises. Better DOA estimation gives a more accurate noise model and vice versa. In contrast, even a little error starts to cause negative feedback in DOA estimation and finally it goes to the fatal error. This paper proposes to introduce a switching scheme, which evaluates the reliabilities of DOA estimates in a stochastic manner and judge whether the noise model should be updated or not into the previously proposed DOA estimator. It is confirmed that the DOA estimator with the proposed scheme can improve its noise robustness even under nonstationary noises.

Session 4pUWa

Underwater Acoustics: Boundary Scattering

Marcia J. Isakson, Chair
Univ. of Texas at Austin, Applied Research Lab., 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

1:30

4pUWa1. Imaging surface roughness with continuous-wave ultrasound reflectometry. Farid G. Mitri, James F. Greenleaf, and Mostafa Fatemi (Mayo Clinic, Dept. of Physio. and Biomedical Eng., Ultrasound Res. Lab., 200 First St., SW, Rochester, MN 55905)

Measurement of surface roughness irregularities is an important indicator of product quality for many nondestructive testing industries. Many techniques exist; however, because of their qualitative, time-consuming and direct-contact modes, it is of some importance to work out new experimental methods and efficient tools for quantitative estimation of surface roughness. Continuous-wave ultrasound reflectometry (CWUR) is presented here as a novel nondestructive modality for imaging and measuring surface roughness in a noncontact mode. In CWUR, voltage variations due to phase shifts in the reflected ultrasound waves are recorded and processed to form an image of surface roughness. An acrylic test block with surface irregularities ranging from 4.22 to 19.05 μm as measured by a coordinate measuring machine (CMM) is scanned by an ultrasound transducer having a diameter of 45 mm, a focal distance of 70 mm, and a central frequency of 3 MHz. It is shown that CWUR technique gives very good agreement with the results obtained through CMM inasmuch as the maximum average percent error is around 11.5%.

1:45

4pUWa2. The effects of roughness on the frequency dependence of specular scattering. Marcia Isakson, R. Abe Yarbrough, and Nicholas Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX 78713, misakson@arlut.utexas.edu)

Specular scattering data were collected at grazing angles from 7 to 77 deg grazing and frequencies from 5 to 50 kHz at the experimental validation of acoustic modeling techniques (EVA) sea test near Isola in October 2006. High resolution microbathymetry data taken using a laser line scan system were also collected. The amplitude and phase of the specular scattered data exhibit a distinctive frequency dependence. This frequency dependence may be due to Bragg scattering from small interface ripples on the ocean bottom indicated by peaks in the measured interface roughness power spectrum. This study simulates the scattering using realizations of interface roughness in a finite element model to determine if the frequency dependence is due solely to scattering or if other mechanisms such as layering are important factors. Results will be compared to analytic approximations such as the Kirchhoff approximation as well as exact integral equations. [Work sponsored by the ONR, Ocean Acoustics.]

2:00

4pUWa3. Finite element modeling of acoustic scattering from rough interfaces. R. Abe Yarbrough and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029)

The finite element method is applied to the problem of acoustic scattering from rough surfaces. This method has the advantage that the surface can have almost any form whereas analytic approximation methods generally require constrained surface roughness parameters. Scattering from a rough pressure release boundary and from a rough interface between two fluid media is considered in two dimensions. The problem domain is truncated with perfectly matched layers. These greatly decrease the number of degrees of freedom in the finite element problem while minimizing unwanted reflections of outbound energy back into the physical domain. The Helmholtz–Kirchhoff integral is used to extend the solution to points outside the modeled domain. Using finite elements as well as an exact integral equation method, solutions are obtained for multiple realizations of the rough scattering surface. The scattering strengths of energy reflected back into the water and of energy penetrating the sea floor in the case of bottom scattering are calculated over a range of scattering angles. The two exact numerical methods show good agreement. Results are compared with those of analytic approximation methods. [Work sponsored by ONR, Ocean Acoustics.]

2:15

4pUWa4. Scattering from an ocean bottom layer using steady-state and transient radiative transfer. Jorge E. Quijano and Lisa M. Zurk (Northwest Electromagnetics and Acoust. Res. Lab., Dept. of Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, zurkl@pdx.edu)

Research on volume scattering from a layered ocean bottom containing random media is usually conducted through methods based on the wave equation. Radiative transfer (RT) theory is an alternative formulation based on the conservation of energy and empirical laws that has the potential of being computationally more efficient and yet can provide considerable insight into the scattering phenomena. The RT theory has been successfully used in electromagnetic remote sensing and characterization of materials using ultrasound. In this work, the RT equation is applied to ocean acoustics to obtain the volume scattering from a layer of sand with flat boundaries overlying a limestone half space. Steady state results are presented to discuss the main processes that contribute to volume scattering and the transformation of energy between shear and longitudinal waves at the boundaries of the layer and at the scatterers. The transient solution of the RT equation as a function of frequency is outlined by introducing an incident pulse of infinitesimal duration that approximates an impulse. The time domain response for a broadband pulse of finite duration is then obtained by convolution with the computed impulse response, which would allow comparison with experimental chirp sonar data. [Research sponsored by the Office of Naval Research, Grant No. N000140510886.]

2:30

4pUWa5. Characterization of the near scattered acoustic vector field. Robert Barton, III and Kevin Smith (Sensors Sonar Dept., Naval Undersea Warfare Ctr., Newport, RI 02841, robert.barton@navy.mil)

In this study, we investigate the properties of the scattered acoustic vector fields generated by simple geometric objects, including the infinite rigid plate, disk, and sphere. Analytical solutions are derived from acoustic target strength scattering models in the near-field region. Of particular interest is the understanding of the characteristics of energy flow of the scattered acoustic vector field in the near- to far-field transition region. We utilize the time and space separable instantaneous active and reactive acoustic intensities to investigate the relative phase properties of the scattered field. Numerical results are presented for the near region scattered acoustic vector field of simple objects in both two and three dimensions.

2:45—2:55 Break

2:55

4pUWa6. An initial look at a continuing shallow water vector sensor acoustic ambient intensity study. David Deveau (Naval Undersea Warfare Ctr., Detachment AUTEC, PSC 1012 Box 701, FPO, AA 34058, david.deveau@autec.navy.mil) and Anthony Lyons (Penn State Univ., University Park, PA 16802)

Deployed in June 2008 off the Coast of Andros Island, Bahamas, seven compact Wilcoxon TV-001 vector sensors have been placed in 15 m of water with the goal of characterizing the ambient noise acoustic intensity with respect to environmental conditions. The study's objective is to better understand how intensity fields describe the shallow water noise environment. The system is designed to gather raw ambient acoustic noise pressure and particle acceleration every hour for a period of one year. The environmental database includes weather and wave information for the local area that can be correlated to the acoustic data. This report will present insight into the overall array development and deployment as well as an initial correlation of intensity fields to the recognized environmental stimulus. Results will be from individual sensors and these same sensors combined into vertical and horizontal arrays utilizing classic linear beamform processing. In addition to gathering data on acoustic ambient noise, data from local passing rain storms, marine mammals, fixed sources, and moving platforms will also be collected and analyzed.

3:10

4pUWa7. Statistics of synthetic aperture sonar image resolution degradation. Shawn F. Johnson, Anthony P. Lyons (Penn State Graduate Program in Acoust., Appl. Res. Lab., 117 Appl. Sci. Bldg., State College, PA 16804), and Douglas A. Abraham (CausaSci LLC, Arlington, VA 22205)

Synthetic aperture sonar (SAS) image statistics can often be characterized by a probability density function with a heavier tail than the expected Rayleigh distribution. The *K*-distribution shape parameter can be used as a metric of non-Rayleighness, with physical ties to both seafloor properties and sonar parameters. Recent results have shown that increasing the resolution cell size, or degrading the resolution of the image, produces images with statistics tending toward Rayleigh (i.e., a higher *K*-distribution shape parameter). In general, a doubling of the resolution cell area results in a doubling of the *K*-distribution shape parameter. A caveat to this generalization is the orientation of the sonar system to any features that may exist on the seafloor (i.e., sand ripples). In such a situation, image statistics may continue to be significantly non-Rayleigh for certain orientations in spite of resolution degradation. SAS images of seafloors with various bottom types and feature orientations collected with an AUV by the Naval Surface Warfare Center Panama City Division have been postprocessed to analyze changes in image statistics as the image resolution is degraded. Results of the image degradation on statistics will be discussed. [Work supported by ONR Grant Nos. N00014-04-1-0013 and 1N00014-06-1-0245, and Code 32.]

3:25

4pUWa8. Bistatic specular reflection by a rigid cone. Philip Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

It is possible to gain some insight into the high-frequency scattering of sound by objects in water by considering the presence or absence of rays constructed from geometric considerations. This presentation concerns the evolution of rays reflected from the sides of a vertical rigid cone for the case of illumination by a plane wave at an arbitrary grazing angle. The grazing angle with respect to a horizontal plane is usually taken to be small. The direction of the reflected rays depends on where the incident ray contacts the cone as specified by an azimuthal angle viewed from above. Define the meridional plane as that plane which contains the incident wave vector and the axis of the cone. Incident rays offset from the meridional plane can be reflected with relatively small vertical components in their wave vector. This analysis has implications for how bistatic hydrophones may be deployed to detect specular glints from cone-shaped objects. There may also be implications for understanding the high-frequency scattering by conical seamounts in deep water. [Research supported by ONR.]

3:40

4pUWa9. Excitation of low-frequency modes of solid cylinders by evanescent and ordinary propagating waves. Aubrey Espana and Phillip L. Marston (Dept. of Phys. and Astronomy, Pullman, WA 99164-2814, aespana81@msn.com)

When using sound to detect objects buried beneath the seafloor, often the acoustic source has a large horizontal stand-off distance. In such situations there is evidence that the incident acoustic wave in sand can have a significant evanescent component. In prior work, organ-pipe modes of water-filled shells were excited in a laboratory simulation, the most significant result being the "double spatial decay rate" effect [Marston *et al.*, J. Acoust. Soc. Am. **122**, 3034 (2007)]. To further understand the coupling by evanescent waves into low-frequency modes of cylinders, backscattering by small solid aluminum cylinders was studied with ordinary-wave illumination. Free-field experiments identified resonances worthy of investigation in an evanescent wave experiment. Several of the features identified have been reproduced in FEM-COMSOL calculations by Williams of APL-UW. For both ordinary and evanescent waves, strong modes were often excited when the cylinder was highly tilted. A further understanding of the modal features was gained by evaluating the temporal variance of the response. Furthermore, the aluminum cylinders also showed an enhanced spatial decay rate when compared to that of the evanescent soundfield. [Work supported by ONR.]

Session 4pUWb

Underwater Acoustics: Forward Scattering and Reverberation

Kevin D. LePage, Chair
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Contributed Papers

4:00

4pUWb1. Rapid simulation of sonar time series. Kevin D. LePage (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia (SP), Italy)

Rapid simulation of element level time series for bistatic sonars requires quality approximations in order to yield accurate results in real time. Here rapid physics-based time series approximations based on closed form expressions for reverberation intensity developed by Harrison [J. Acoust. Soc. Am. **114**, 2744 (2003)] are developed for boundary reverberation and target scattering in shallow water waveguides. The simulations include Doppler and waveguide angle and time dispersion effects, and since the simulations are at the element level, they can be processed like real data. Simulations are compared to reverberation predictions obtained with more accurate but slower codes showing good agreement.

4:15

4pUWb2. Higher moment estimation for reverberation simulation. Kevin D. LePage (NATO Undersea Res. Ctr., Viale San Bartolomeo 400, 19126 La Spezia (SP), Italy)

In recent work the second, third, and fourth spatial moments for exponentially distributed roughness with either Gaussian or von Karman spatial spectra have been presented by the author. In this talk results for the second, third, and fourth moments of reverberation pressure for perturbation theory scattering from surfaces with von Karman spectra are obtained using numerical quadrature. Results are compared to closed form expressions for the second and fourth moments obtained for Gaussian correlation functions. Methods to extend these results to the class of chi-square distributed roughness heights of arbitrary order, which encompasses both exponential and Gaussian height distributions, are also presented.

4:30

4pUWb3. Examination of loss mechanisms for a rough bottom Pekeris waveguide using a two-way coupled mode model. Steven Stotts, Robert Koch, and David Knobles (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78752)

A benchmark scattered-field solution for a two-dimensional rough-bottom Pekeris ocean waveguide has been generated from a two-way coupled-mode model with normal modes evaluated in the complex horizontal wave-number plane. The received level time series coupled-mode solution for a range monostatic source/receiver geometry with a 250 Hz center frequency, 20 Hz band, Gaussian pulse was shown previously to be consistent with a single scatter Born approximation solution [Stotts *et al.*, J. Acoust. Soc. Am. **122**, 3075 (2007)]. A new result for a 1 kHz center frequency, 60 Hz band, pulse deviates significantly from the Born approximation solution. Alternative approaches, obtained with the Born approximation but including the additional propagation loss for Kirchhoff scattering [E. I. Thorsos, J. Acoust. Soc. Am. **83**, 78–92 (1988)] with and without second order small slope approximation effects are examined. The solutions at 1 kHz with these alternative methods are compared to the coupled-mode solution. The forward scattered fields propagated over finite length roughness segments of varying lengths are analyzed to understand the role of mode-mode interactions in the forward propagation loss mechanisms and for comparison with the Kirchhoff scattering loss description.

4:45

4pUWb4. Time-domain solutions for Rayleigh and Stoneley waves using the single-scattering parabolic equation method. Adam M. Metzler, William L. Siegmann (Rensselaer Polytechnic Inst., Troy, NY 12180), Michael D. Collins (Naval Res. Lab., Washington, DC 20375), Robert A. Zingarelli, and Stanley A. Chin-Bing (Naval Res. Lab., Stennis Space Ctr., MS 39529)

The parabolic equation method implemented with the single-scattering correction accurately handles range-dependent environments in elastic layered media. Interfaces between elastic media may be treated efficiently by subdividing into a series of two or more single-scattering problems [Küsel *et al.*, J. Acoust. Soc. Am. **121**, 808–813 (2007)]. In addition to environmental waveguide parameters, the procedure uses several computational parameters. The impacts of the number of interfacial scattering problems, an iteration scheme convergence parameter, and the number of iterations for convergence are shown on the accuracy and efficiency of the method. In particular, selection criteria for these parameters are developed. Fourier transforms and syntheses generate time-domain solutions for seismic applications of interest. Examples for model waveguides show features of Rayleigh and Stoneley wave propagation, and comparisons with solutions from other methods are shown. [Work supported by the ONR.]

5:00

4pUWb5. Monte Carlo simulation of rough surface scattering using a modified pseudospectral time-domain method. Yonghoon Ha, Keunwha Lee, and Woojae Seong (Dept. of Ocean Eng., Seoul Nat'l Univ., San 56-1, Sillim-dong, Seoul 151-744, Korea)

A pseudospectral time-domain (PSTD) method using a surface flattening transformation and image method is applied to a randomly rough pressure-release surface scattering problem. Above, the PSTD method efficiently avoids the Gibbs phenomenon appearing in the conventional PSTD method when treating irregular boundaries with large impedance contrast. The pressure-release surface is generated by Pierson–Moskowitz spectrum and Monte Carlo method is used to calculate the statistical properties of the rough surface. The Monte Carlo results for elemental scattering area will be investigated by comparing with Chapman–Harris formula.

5:15

4pUWb6. Three-dimensional spatial coherence measurements: Vertical, longitudinal/transverse horizontal coherence. Lin Wan, Ji-Xun Zhou, Peter Rogers (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, lin.wan@gatech.edu), and David Knobles (The Univ. of Texas at Austin, Austin, TX 78713)

In the Shallow Water '06 experiment, two L-shape arrays were deployed. Two horizontal components of these arrays were laid on the sea bottom. One was exactly perpendicular to the direction of sound propagation. The other was exactly parallel to the direction of sound propagation. This configuration offered an opportunity to directly measure the vertical, longitudinal horizontal, and transverse horizontal coherence. The results of spatial coherence were averaged over different pairs of hydrophones and over a frequency bandwidth of 100 Hz. The vertical coherence showed receiver depth dependence. When the source and the receivers were located below the thermocline, both the vertical and longitudinal hori-

zontal coherence lengths (in units of wavelength) increased with increasing range and frequency. The longitudinal horizontal coherence length was much larger than the vertical coherence length. These results were similar to the predictions by Smith's model [J. Acoust. Soc. Am. **60**, 305–310 (1976)] with a frequency dependent bottom reflection loss [Zhou, Chin. Phys. **1**, 494–504 (1981)]. The transverse horizontal coherence length/wavelength decreased with frequency. When the source depth was within the thermocline, the transverse horizontal coherence lengths exhibited weak range dependence in the 100–300 Hz range and was larger than 40 wavelengths. [Work supported by ONR.]

5:30

4pUWb7. Model for scattered field from a vertically extended target in a range-dependent ocean waveguide. Elizabeth Küsel and Purnima Ratilal (Dept. Elelectrical and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, kusele@alum.rpi.edu)

A model for the scattered field from a vertically extended target in a range-dependent ocean waveguide is developed by application of Green's theorem. The model is implemented with the parabolic equation (PE) method for a vertically extended pressure-release cylinder, such as the BBN target. The local scattered field at the surface of the target is approximated by the sum of Hankel functions. The total scattered field is calculated using Kirchhoff's integral approximation to Green's theorem. Results of calculations for a Pekeris waveguide show good agreement with the Ingenito normal-mode-based model for target scattering in a waveguide. A limitation with the Ingenito scattering model is that the target has to be located within an isospeed layer. The PE-based scattering model developed here, on the other hand, is able to take into account sound speed variations along the vertical extent of the target. We show that using the Ingenito model in a layered waveguide with the vertically extended target can lead to inaccuracies in estimating the scattered field. Illustrative examples are also provided for range-dependent environments.

5:45

4pUWb8. Atlantic herring low-frequency target strength and abundance estimation: Ocean acoustic waveguide remote sensing (OAWRS) 2006 Experiment in the Gulf of Maine. Zheng Gong, Mark Andrews, Daniel Cocuzzo, Saumitro Dasgupta, Purnima Ratilal (Dept. of Elec. and Comput. Eng., Northeastern Univ., Boston, MA 02115), Srinivasan Jagnnathan, Deanelle Symonds, Ioannis Bertsatos, Tianrun Chen, Nicholas Makris (MIT, Cambridge, MA 02139), Redwood Nero (Naval Res. Lab., Stennis Space Ctr., MS 39529), Hector Pena, Ruben Patel, Olav Rune Godoe (Inst. of Marine Res., Nordnes, N-5817 Bergen, Norway), and J. Michael Jech (Northeast Fisheries Sci. Ctr., Woods Hole, MA 02543)

The mean low-frequency target strength (TS) of spawning Atlantic herring populations in the Gulf of Maine is estimated from the experimental data acquired during September–October 2006 near the northern flank of Georges Bank. A low-frequency OAWRS system with an instantaneous imaging diameter of 100 km was deployed to provide spatially unaliased imaging of fish populations over wide areas. The OAWRS system's scattering strength measurements are calibrated with areal fish population density estimates obtained from concurrent localized line-transect measurements with several conventional fish finding sonars (CFFSs). Trawl sampling at selected locations enables the identification of the imaged species. The mean TS estimates of herring individuals exhibits significant variation over OAWRS operating frequency range, in accordance with the results from a resonant scattering model for swimbladder-bearing fish. The neutral buoyancy depth of herring and the species composition in the imaged population is inferred by comparing the measured TS with those derived from the model. Our analysis indicates that the herring population has a neutral buoyancy depth of between 70 and 90 m and is therefore negatively buoyant between 120 and 180 m water depth at which it is commonly found. The herring populations instantaneously imaged with OAWRS often exceeds 200×10^6, of which over 150×10^6 individuals can be organized into a large shoal.