Session 3aAA

Architectural Acoustics, Noise, Physical Acoustics, Signal Processing in Acoustics and Underwater Acoustics: A Celebration of the Life in Acoustics of Richard H. Bolt

William J. Cavanaugh, Chair
Cavanaugh Tocci Associates, Inc., 327F Boston Post Road, Sudbury, Massachusetts 01776

Chair's Introduction—8:30

Invited Papers

8:35

3aAA1. Richard H. Bolt: Scientist, educator and friend. Leo L. Beranek (975 Memorial Dr., Ste. 804, Cambridge, MA 02138-5755)

The author will discuss his close relation to Dick Bolt, starting with their first meeting in 1938. This will include Bolt's doctoral years at the University of California, his National Research Fellowship in Physics at MIT, his contributions to the World War II effort, his tenured appointment, guidance of students and formation of the Acoustical Laboratory at MIT, and his co-founding of and subsequent participation in Bolt Beranek and Newman. Dick's principal contributions to the field of acoustics, to education and to the Acoustical Society of America will be presented up until 1957, after which his talents were largely taken over by national missions, which are mentioned. Richard H. Bolt was an inspiring educator, researcher and organizer. He truly was a national treasure.

9:00

3aAA2. Richard H. Bolt introduced me to acoustics, and I am still engaged. Ira Dyer (Dept. of Ocean Eng., MIT., Cambridge, MA 02139)

I arrived at the MIT Acoustics Laboratory in 1948, and quickly learned from Dick Bolt that U.S. submarines needed coatings to reduce vulnerability via active sonar detection. All who remember Dick will be familiar with his next step: He brought me to Robert Blizzard and A. Wilson Nolle, both then at the Lab, from whom I learned in detail how to measure and then how to understand the dynamic moduli of rubber-like materials. My path, of course, widened to include medical acoustics (under the leadership of Theodore Hueter) and scattering (Phillip Morse and Uno Ingard). Fast-forward to the present. Rubber-like materials are not only in use as submarine coatings to avoid detection by active sonars, but also to reduce noise radiated by submarines. Further, such materials cover submarine acoustic arrays to reduce flow–noise interference with signals received by such arrays. The problem that most engages me today is the stochastic nature of sound propagation in the ocean, a problem that has the practical consequence of degraded sonar performance. In the spirit of Dick, I will present snippets of my current work, with the hope of painting the big picture that he always did so well.

9:25

3aAA3. Richard H. Bolt's work on theoretical aspects of room acoustics. Richard H. Lyon (RH Lyon Corp, Cambridge, MA 02138, rhlyon@lyoncorp.com)

From about 1940 to 1950 Dick Bolt undertook a number of studies of the transmission of sound in a room. These were mostly theoretical and appear to have had two sources of inspiration: the work of Wente at Bell Laboratories in the 1930s, and Philip Morse's work on the eigenfunctions and eigenvalues of idealized rooms in the early 1940s. In 1944, the Reviews of Modern Physics published in its April issue a single paper—that of Dick Bolt and Phil Morse, titled "Sound waves in rooms." This *tour de force* took the viewpoint that a deterministic calculation of the modes of an idealized room could reveal important features of the acoustics of less ideal spaces. The relations between wall impedance and modal damping for axial, tangential, and oblique modes and the construction of direct fields from modal expansions are worthy of special note. Perturbation analysis is used to couple idealized modes as a first attempt to deal with "real" rooms and the transition to ergodic behavior. This work then led Bolt to a number of studies of the irregularity of spacing of modal resonance frequencies, which had some influence, in turn, on models of modal distributions in Statistical Energy Analysis.

9:50–10:10 Break

10:10

3aAA4. Acoustics and its relation to language: The influence of Dick Bolt. Kenneth N. Stevens (Res. Lab. of Electronics and Dept. of Elec. Eng. and Computer Sci., MIT, Cambridge, MA 02139)

Under the mentoring of Dick Bolt, and the stimulation he provided in the Acoustics Laboratory at MIT, many students were exposed to a range of topics in acoustics, including mechanisms of sound generation, radiation, and propagation, noise control, acoustics of resonators and rooms, human responses to sound, and speech perception under various adverse conditions. As someone who became interested in speech communication, I have recognized that this kind of quantitative background in acoustics is an important requirement for developing models of how humans produce speech, how they perceive and understand speech, and how children acquire these skills. Speech production involves sound sources produced by a nonlinear mechanical system and by noise arising from turbulent airflow. Sound is propagated in a vocal tract with yielding walls, and acoustic coupling is introduced by lossy resonators attached to the vocal tract, including the trachea and the nasal cavity. These acoustic principles of sound generation create an inventory of sound types that give rise to distinctive responses in the ears and brains of listeners. The solid grounding in acoustics provided by Dick Bolt and his leadership have helped in the formation of this linkage between acoustics, speech physiology, linguistics, and human perception.

10:35

3aAA5. Richard H. Bolt—Mentor and colleague. Ewart A. Wetherill (Shen, Milsom & Wilke/Paoletti, 649 Mission, San Francisco, CA 94105, rwetherill@sf.smwinc.com)

Of his many accomplishments in acoustics, perhaps one of Richard H. Bolt's greatest legacies will prove to be his contribution, both directly and through his teaching, to everyday hearing conditions in buildings. In a discipline that attempts to bridge the technical and cultural gap between a pure science and the pragmatic and often-haphazard process of building design, he combined a deep understanding of both professions with an ability to communicate complex ideas that is reminiscent of Wallace Clement Sabine. His welcoming enthusiasm and humility enabled him to attract and to work well with people of complementary talents, in both theoretical research and the gritty details of a consulting practice, as well as to envision the potential of still-unexplored subjects. A logical outcome of this combination was the profoundly influential technical group known as BBN, whose pioneering integration of acoustics with building technology is echoed by many consulting groups throughout the world.

11:00–11:40

Panel Discussion

WEDNESDAY MORNING, 30 APRIL 2003 ROOM 201, 7:55 A.M. TO 12:05 P.M.

Session 3aAO

Acoustical Oceanography, Animal Bioaoustics and Underwater Acoustics: Bioacoustic Resonance Spectroscopy

Orest I. Diachok, Chair

Naval Research Laboratory, Code 7420, 4555 Overlook Avenue Southwest, Washington, D.C. 20375-0002

Chair's Introduction—7:55

Invited Papers

8:00

3aAO1. Acoustic swimbladder resonance spectroscopy: Fundamentals in scattering theory. David T. I. Francis (Dept. of Electron., Elec. and Computer Eng., Univ. of Birmingham, Birmingham B15 2TT, UK) and Kenneth G. Foote (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

A history of the physics of acoustic resonance is given. The primary, low-frequency, resonant scattering model for air bubbles in water [Minnaert (1933)] is reviewed. Subsequent applications to swimbladdered fish, including models by Andreeva (1964), Love (1978), and Feuillade and Nero (1998), among others, are developed. Reference is made to exemplary measurements of backscattering by Holliday (1972) and Loevik and Hovem (1979), and of forward scattering, or absorption, by Weston (1967) and Diachok (2000), among others. High-frequency resonances are also described, with presentation of both analytical and numerical results for the immersed air bubble. Comparison of these validates the numerical, boundary-element method (BEM). The BEM allows high-frequency resonances to be studied for swimbladders of realistic shapes under pressure and for typical wave-number-swimbladder length products of order 10–40. Implications of high-frequency swimbladder resonance for auditory function in fish are mentioned. [Work supported by ONR.]

9:00

3aAO2. High-frequency acoustic scattering from gas-bearing zooplankton. Andone C. Lavery, Timothy K. Stanton, Peter H. Wiebe (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543), Mark C. Benfield (Louisiana State Univ., Baton Rouge, LA 70803), and Charles H. Greene (Cornell Univ., Ithaca, NY 14853)

High-frequency acoustic scattering is well-suited to the synoptic investigation of marine organisms that inhabit the water-column, such as zooplankton and fish. However, the scattering characteristics of the organisms can be highly complex, and one must look for ways to distinguish one type of organism from another when interpreting the received echoes. From an acoustic scattering perspective,

zooplankton generally fall into one of three categories: gas-bearing, fluid-like, or elastic-shelled. Scattering models, of different levels of sophistication are available for all three types of zooplankton. Gas-bearing zooplankton, unlike the other zooplankton categories, have a strong scattering resonance, which can be exploited for their identification. Scattering from gas-bearing zooplankton at frequencies close to the resonance frequency is much stronger than scattering from other zooplankton. Thus, in this frequency region, acoustic scatter from a small number of gas-bearing zooplankton can overwhelm that from a far larger number of fluid-like or elastic-shelled zooplankton. A series of zooplankton surveys of the Gulf of Maine has recently been completed in which a multi-frequency towed instrument platform, BIOMAPER-II, was employed. The strong scattering resonance of gas-bearing zooplankton, together with scattering models and ground-truthing, was exploited in order to determine regions with high densities of gas-bearing zooplankton.

9:20

3aAO3. Potential effect of resonant scattering from multiple swimbladders on audition in juvenile fish. Mardi C. Hastings (ONR 342, BCT-1, 800 N. Quincy St., Arlington, VA 22217, hastinm@onr.navy.mil)

The swimbladder, a gas-filled chamber in the abdominal cavity of most bony fishes, is a hydrostatic organ that enables fish to maintain neutral buoyancy; however, it also responds to acoustic pressure and radiates a secondary acoustic field that enhances detection capability of the inner ear. Recent experiments have indicated that resonant response of the swimbladder may control the auditory bandwidth in at least four species of fish [Hastings *et al.*, J. Acoust. Soc. Am. **110**, 2640 (2001)]. The auditory bandwidths of these fishes, however, do not change appreciably while they grow even though the resonance frequency of the swimbladder decreases with increasing body length. Results of an analysis inspired by Feiullade *et al.* [J. Acoust. Soc. Am. **112**, 2206 (2002)] show that the downward shift and broadening associated with resonance of the aggregate scattered field from multiple fish is perhaps sufficient enough to account for this discrepancy. Effects of resonant characteristics of a single swimbladder, fish length, and number of fish on the changes in the collective scattered field are presented. Thus the resonant scattered field created by relatively large schools of juvenile fish may enhance their auditory capability.

9:40

3aAO4. Experimental measurements of lung resonant frequencies in a bottlenose dolphin (*Tursiops truncatus*) and white whale (*Delphinapterus leucas*). James J. Finneran (U.S. Navy Marine Mammal Prog., Space and Naval Warfare Systems Ctr., San Diego, Code 2351, 53560 Hull St., San Diego, CA 92152)

An acoustic backscatter technique was used to estimate *in vivo* whole-lung resonant frequencies in a bottlenose dolphin (Tursiops truncatus) and a white whale (Delphinapterus leucas). Subjects were trained to submerge and position themselves near an underwater sound projector and a receiving hydrophone. Acoustic pressure measurements were made near the subjects' lungs while insonified with pure tones at frequencies from 16 to 100 Hz. Whole-lung resonant frequencies were estimated by comparing pressures measured near the subjects' lungs to those measured from the same location without the subject present. Experimentally measured resonant frequencies and damping ratios were much higher than those predicted using equivalent volume spherical air bubble models. The experimental technique, data analysis method, and discrepancy between the observed and predicted values will be discussed. The potential effects of depth on the resonance frequencies will also be discussed.

10:00–10:10 Break

10:10

3aAO5. The influence of adjacent structure on the response of insonified fish and marine mammal resonant cavities. Joel Garrelick (Applied Physical Sciences, Inc., 2 State St., Ste. 300, New London, CT 06320)

It has long been known that the acoustic response of, and scattering from, finite elastic structures and cavities may be enhanced at their submerged natural frequencies, that is, at resonance. Fish and marine mammal cavities, viz., air bladders and lungs, respectively, are prime examples. In either case, classic analyses represent the cavity as being in an otherwise free field and spherical in geometry, initially as well as in its deformed state. Such simplifying assumptions may be invalidated by the presence of adjacent structure, whether or not such structure is transparent to the incident wave. This is especially the case for local response functions, namely stresses and strains in surrounding tissue, as opposed to say the acoustically scattered return. A strong sensitivity of these response functions to the assumed geometry compounds the problem. These issues are analytically examined in this paper for the example of a plane acoustic wave incident on a membrane-sheathed cavity located adjacent to structure of prescribed high or low impedance. Predictions for the scattered acoustic field and cavity stress/strains are compared to classic free field values. Enhanced levels are computed and interpreted in terms of mode coupling to higher order membrane/cavity vibration modes.

10:30

3aAO6. Measurements and bioeffects of resonant gas bodies *in vivo*. Diane Dalecki (Dept. of Biomed. Eng., Univ. of Rochester, Rochester, NY 14627, dalecki@bme.rochester.edu)

The response of resonant gas bodies *in vivo* (such as the gas filled lung and intestine of mammals) to exposure to low frequency underwater sound was characterized through a series of investigations. Measurements were performed in the laboratory using a specially designed acoustic exposure system, capable of generating maximum acoustic fields of ~200 dB *re* 1 μPa for frequencies

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spanning 100–2500 Hz. Both acoustic scattering and pulse-echo ultrasound techniques were used to characterize the response of gas bodies to underwater sound exposure and to determine the resonance frequencies of gas bodies in mammalian tissues *in vivo*. A series of investigations has demonstrated that the effects of low frequency (~ 100–2500 Hz) underwater sound can be significant in and near tissues that contain resonant gas bodies. For example, mice exposed to underwater sound at the resonance frequency of their lung exhibited lung damage and the extent of tissue damage increased with increasing pressure amplitude. Similar types of investigations were performed with mammalian lungs of various sizes and with intestinal gas *in vivo*.

Contributed Papers

10:50

3aAO7. Observations of frequency shift associated with schooling fish. Orest Diachok (Naval Res. Lab., Washington, DC 20375, orest@wave.nrl.navy.mil)

The number of sardines per school, N, is nominally 10 000 and the separation between sardines in school, s, is nominally 1 fish length, $L \cdot s$ is much smaller than the wavelength at f (the resonance frequency of individuals), which suggests that schools may exhibit acoustic properties of bubble clouds. Long-term, broadband transmission loss measurements at a shallow-water site in the Gulf of Lion revealed absorption lines due to dispersed sardines at 1.3 kHz at 20 m at night and 2.7 kHz at 65 m at dawn. Temporal changes in observed values of f were consistent with concurrent echo sounder observations of the vertical migration of sardines, and theoretical computations based on laboratory measurements of swim bladder dimensions. The measured resonance frequency of sardines in schools during daytime, 1.7 kHz at 65 m, was 0.6f at the same depth at dawn. The observed frequency shift is consistent with a hybrid model of the fundamental resonance frequency of a bubble cloud, which is based on theories developed by Feuillade, Nero, and Love (1996) and dAgostino and Brennan (1988), and $s=0.8\ L$ and $N=5000$ fish per school. [Work supported by ONR.]

11:05

3aAO8. Backscattering and extinction cross sections of two swimbladdered fishes at the lowest resonance, as modeled by the boundary-element method. Kenneth G. Foote (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543) and David T. I. Francis (Univ. of Birmingham, Birmingham B15 2TT, UK)

The boundary-element method has been applied to backscattering and extinction of sound by swimbladdered fish at the lowest, breathing-mode resonance. Corresponding cross sections have been computed for specimens of two representative kinds of swimbladder-bearing fish, namely physostomes and physoclists, which, respectively, possess and lack an external duct. The respective fishes are herring (*Clupea harengus*) and pollack (*Pollachius pollachius*), for which swimbladder morphometric data are available. The depth dependences of the cross sections are computed over the range 0–500 m. Comparisons are made with measurements and other modeled results for a number of species. [Work supported by ONR.]

11:20

3aAO9. Target strength and density structure of Hawaiian mesopelagic boundary community patches. Kelly Benoit-Bird and Whitlow Au (Hawaii Inst. of Marine Biol., P.O. Box 1106, Kailua, HI 96734, benoit@hawaii.edu)

A broadband sonar system and digital camera with strobe lights were mounted on a vertically profiling frame with a depth sensor. The target strengths and densities of animals within individual mesopelagic boundary community patches were investigated as a function of depth. Simultaneous echosounder surveys permitted comparison of density estimates from echo-energy integration and echo-highlight counting. Target strength values suggest nearshore boundary community animals are primarily myctophid fishes which was confirmed by preliminary photographic evidence. Target strength varied significantly as a function of distance from the shoreline and time. Echo-energy integration estimates of density made with these revised target strengths compare well with those made with echo highlight counting. These density measures show that previous density estimates were too low but do not change the conclusions of these studies. Vertical microstructure in density was apparent but animal size and compositional structure was not evident within a patch. Patch edges were abrupt, with no differences in the density or target strength from patch interiors. These edges were generally straight, with a sharp drop in density to the background density of zero. Estimates of animal size as a function of time provide information about the diel migration patterns of these mesopelagic animals.

11:35

3aAO10. Inversion of the depth, thickness, and absorption coefficient of a layer of fish (anchovies) from transmission loss measurements in the Yellow Sea. Orest Diachok and Stephen Wales (Naval Res. Lab., Washington, DC 20375, orest@wave.nrl.navy.mil)

Estimates of bioacoustic parameters of fish (anchovies) and geoacoustic parameters of the bottom were simultaneously inverted from the TL measurements of Qiu *et al.* (1999) in the Yellow Sea. Replica fields were calculated with BIO-C-SNAP, a C-SNAP based normal mode model, which permits inclusion of bio-absorption layers. The inversion was based on minimizing the rms difference, D, between measured and calculated values of TL at multiple source and receiver depths and ranges, and involved a simultaneous search for bio-layer depth (d), bio-layer thickness, bio-alpha (a), geo-sound speed, and geo-alpha. The resultant values of D were extremely small (1.9 dB at 1.35 kHz). By contrast, inversion calculations, which assumed that all excess attenuation at this site was due to the bottom, resulted in unacceptably large values of D (9.5 dB at 1.35 kHz). The inverted value of d, 6.8 m, was consistent with laboratory measurements of the resonance frequencies of 10-cm anchovies (1.35 kHz at 6 m), the dominant species in the Yellow Sea. The inverted value of a was consistent with previously reported number densities of anchovies in the Yellow Sea. Inverted geoacoustic parameters were within previously reported bounds. [Work supported by ONR.]

11:50

3aAO11. Studies on geometrical backscattering models of marine bodies. Anil Kumar C. Parameswaran, Sajith N. Pai, N. Soniraj, P. R. Saseendran Pillai, James Kurian, Supriya M. Hariharan, C. Madhavan, and T. K. Mani (Cochin Univ. of Sci. and Technol., Cochin 682022, India)

The target strength of marine bodies depends on two components—one of them is easy to measure and a good relationship can be established with the target strength value, while the other presents a higher variability that no method can help to reduce, which include the orientation of the fish and a range of environmental conditions and a set of biological facts. Studies on physical models and aspects are expected to provide a clear insight into the issues relating to the target strength variability. Such physical models are developed by converting the physiological shape of the fish into standard and simple geometrical shapes. Data obtained from some of the commercially important species, individually positioned at the center of the acoustic beam, 3 m from the transducer in a test facility were used for the computation of target strength. The target strength value obtained from these reference targets is an indicator of the model performance. Mathematical description of the scattering by some of the species and subsequent comparison with laboratory data have demonstrated that the scattered level by an individual due to a single ping, strongly depends upon size, shape, frequency, material properties, and orientation. Perhaps one of the most notable peculiarities of this work is the simplicity of the approximation and the close agreement between the real world value and the model solution.

Session 3aBB

Biomedical Ultrasound/Bioresponse to Vibration: Audible-Frequency Medical Diagnostic Methods

Thomas J. Royston, Chair

Department of Mechanical Engineering, University of Illinois at Chicago, Mail Code 251, 842 West Taylor Street, Chicago, Illinois 60607-7022

Chair's Introduction—8:00

Invited Papers

8:05

3aBB1. Audible-frequency methods in assessment of soft tissues. Armen P. Sarvazyan (Artann Labs., 1753 Linvale-Harbourton Rd., Lambertville, NJ 08530)

One of the reasons why MHz-range ultrasonic compressional waves have become a powerful tool for medical diagnostics is that their wavelength is of the order of a millimeter, adequate for imaging an anatomical structural. Other types of acoustic waves that can be generated in soft tissue, such as shear and surface waves, are two–three orders of magnitude slower and, respectively, have a millimeter range wavelength in the audible-frequency range. These waves can be effectively used for imaging and soft tissue characterization in terms of the shear elasticity modulus. This paper is a summary of the studies on applications of the surface and shear waves for diagnostic assessment of soft tissue elasticity and dynamic viscosity performed by the author and his coworkers over three decades. A number of techniques for measuring shear elastic moduli of soft tissue using both wave propagation and the acoustic impedance methods are described. Audible-frequency methods and devices for assessment of skin in dermatological applications, for the detection of breast cancer, for imaging of brain and muscle tissues are analyzed. Various embodiments of the Shear Wave Imaging modality are considered. The reasons for a limited success of the audible-frequency methods despite their great potential in ultrasonic medical imaging and diagnostics are discussed.

8:30

3aBB2. Vibro-acoustography: The sound of tissue. James Greenleaf and Mostafa Fatemi (Mayo Clinic, 200 First St. SW, Rochester, MN 55905)

Palpation is routinely used for detecting pathology using the stiffness of the tissue and is more than 2000 years old. Palpation is subjective and limited to individual experience and to the accessibility of the tissue region to touch. Noninvasive images of elastic modulus may be useful to distinguish tissues and pathologic processes based on mechanical properties such as elastic modulus. Static, quasistatic, or cyclic stresses have been applied while strains have been measured using ultrasound or MRI. The related elastic modulus has been estimated from visco-elastic models of tissue mechanics. Recently we have developed a new ultrasound technique that produces speckle free images related to both tissue stiffness and reflectivity. This method, termed Ultrasound Stimulated Vibro-acoustography [Science **280**, 83–85 (1998); PNAS **96**, 6603–6608 (1999)], uses ultrasound radiation pressure to produce sound vibrations from a small region of the tissue that depend on the scattering and elastic characteristics of the tissue. The method can detect microcalcification within breasts, and promises to provide high-quality images of calcification within the arteries. Vibro-acoustography can detect mechanical defects in prostheses such as artificial mitral and aortic valves. The method may also be used in nondestructive evaluation. [Work supported by HL 61451, DBI 7980-4.]

8:55

3aBB3. Supersonic imaging of elasticity. Mathias Fink, Michael Tanter, and Jeremy Bercoff (Laboratoire Ondes et Acoustique, 10 rue Vauquelin, 75005 Paris, France)

This paper shows the new possibilities of supersonic imaging of elasticity. It combines ultra-high speed ultrasonic imaging of tissues with low frequency supersonic shear sources induced in the body by radiation pressure. We have built an Ultrafast Scanner providing up to 5000 frames/s and using conventional arrays linked to a new beamformer. We will show how this scanner is able to follow in real time tissue displacements induced by the transient propagation of LF shear wave in the human body. From the spatio-temporal evolution of the displacements, a shear elasticity map is constructed using a local inversion algorithm. In order to obtain unbiased shear elasticity map, different configurations of shear sources induced by radiation pressure of focused transducer arrays are used. A very interesting configuration that induces quasi plane shear waves will be described. It used a shear source that moves at supersonic velocities, and that is created by using a very peculiar beam forming in the transmit mode. *In vitro* and *in vivo* results will be presented that demonstrate the interest of this new transient elastographic technique.

9:20

3aBB4. Vibration amplitude sonoelastography lesion imaging using low-frequency audible vibration. Lawrence Taylor (Biomed. Eng. Dept., Univ. of Rochester, P.O. Box 270168, Rochester, NY 14627) and Kevin Parker (Univ. of Rochester, Rochester, NY 14627)

Sonoelastography or vibration amplitude imaging is an ultrasound imaging technique in which low-amplitude, low-frequency shear waves, less than 0.1-mm displacement and 1-kHz frequency, are propagated deep into tissue, while real time Doppler techniques are used to image the resulting vibration pattern. Finite-element studies and experiments on tissue-mimicking phantoms verify that a discrete hard inhomogeneity present within a larger region of soft tissue will cause a decrease in the vibration field at its location. This forms the basis for tumor detection using sonoelastography. Real time relative imaging of the vibration field is possible because a vibrating particle will phase modulate an ultrasound signal. The particle's amplitude is directly proportional to the spectral spread of the reflected Doppler echo. Real time estimation of the variance of the Doppler power spectrum at each pixel allows the vibration field to be imaged. Results are shown for phantom lesions, thermal lesions, and 3-D *in vitro* and 2-D *in vivo* prostate cancer. MRI and whole mount histology is used to validate the system accuracy.

9:45

3aBB5. Acoustic detection of pneumothorax. Hansen A. Mansy (Biomed. Acoust. Res. Group, Rush Medical College, 1725 W. Harrison St., Ste. 946, Chicago, IL 60612, hmansy@rush.edu), Thomas J. Royston (Univ. of Illinois at Chicago, Chicago, IL), Robert A. Balk, and Richard H. Sandler (Biomed. Acoust. Res. Group, Rush Medical College, Chicago, IL 60612)

This study aims at investigating the feasibility of using low-frequency (<2000 Hz) acoustic methods for medical diagnosis. Several candidate methods of pneumothorax detection were tested in dogs. In the first approach, broadband acoustic signals were introduced into the trachea during end-expiration and transmitted waves were measured at the chest surface. Pneumothorax was found to consistently decrease pulmonary acoustic transmission in the 200–1200-Hz frequency band, while less change was observed at lower frequencies ($p<0.0001$). The ratio of acoustic energy between low (<220 Hz) and mid (550–770 Hz) frequency bands was significantly different in the control (healthy) and pneumothorax states ($p<0.0001$). The second approach measured breath sounds in the absence of an external acoustic input. Pneumothorax was found to be associated with a preferential reduction of sound amplitude in the 200- to 700-Hz range, and a decrease of sound amplitude variation (in the 300 to 600-Hz band) during the respiration cycle ($p<0.01$ for each). Finally, chest percussion was implemented. Pneumothorax changed the frequency and decay rate of percussive sounds. These results imply that certain medical conditions may be reliably detected using appropriate acoustic measurements and analysis. [Work supported by NIH/NHLBI #R44HL61108.]

10:10–10:20 Break

10:20

3aBB6. Passive acoustic monitoring of human physiology during activity indicates health and performance of soldiers and firefighters. Michael V. Scanlon (US Army Res. Lab., AMSRL-SE-SA, 2800 Powder Mill Rd., Adelphi, MD 20783-1197, mscanlon@arl.army.mil)

The Army Research Laboratory has developed a unique gel-coupled acoustic physiological monitoring sensor that has acoustic impedance properties similar to the skin. This facilitates the transmission of body sounds into the sensor pad, yet significantly repels ambient airborne noises due to an impedance mismatch. The sensor's sensitivity and bandwidth produce excellent signatures for detection and spectral analysis of diverse physiological events. Acoustic signal processing detects heartbeats, breaths, wheezes, coughs, blood pressure, activity, motion, and voice for communication and automatic speech recognition. The health and performance of soldiers, firefighters, and other first responders in strenuous and hazardous environments can be continuously and remotely monitored with body-worn acoustic sensors. Comfortable acoustic sensors can be in a helmet or in a strap around the neck, chest, and wrist. Noise-canceling sensor arrays help remove out-of-phase motion noise and enhance covariant physiology by using two acoustic sensors on the front sides of the neck and two additional acoustic sensors on each wrist. Pulse wave transit time between neck and wrist acoustic sensors will indicate systolic blood pressure. Larger torso-sized arrays can be used to acoustically inspect the lungs and heart, or built into beds for sleep monitoring. Acoustics is an excellent input for sensor fusion.

10:45

3aBB7. Bone and tissue conduction of high intensity acoustic energy to the human cochlea. Richard McKinley (Air Force Res. Lab., AFRL/HECB, 2610 Seventh St., Wright-Patterson Air Force Base, OH 45433-7901, richard.mckinley@wpafb.af.mil), Armand Dancer (French German Res. Inst., St. Louis, France), and Henning von Gierke (Air Force Res. Lab. (Emeritus), AFRL/HECB, OH 45433-7901)

Noise fields near operating high performance fighter aircraft range from 140 to 150 dB overall sound pressure level. Much of the acoustic energy which is transmitted to the cochlea in a well protected ear (earplugs and earmuffs) arrives by tissue and bone conduction pathways. A better understanding of this transmission of acoustic energy by these biological materials could lead to improved protection and/or methods of actively attenuating this acoustic energy in the cochlea. Associated issues to be discussed are the possible damaging levels of bone/tissue conducted acoustic energy and the actual bone/tissue attenuation. This paper will describe a series of experiments in a recent investigation of bone and tissue conduction of acoustic energy over a broad band of frequencies and a look at the initial results of the first of the studies. [Work supported in part by the Air Force Office of Scientific Research.]

Contributed Papers

11:10

3aBB8. A vibration model for frequency analysis of arterial tubes with tissue. Xiaoming Zhang, Mostafa Fatemi, and James F. Greenleaf (Ultrasound Res. Lab., Mayo Clinic, 200 First St. SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

Vibro-acoustography is a new noncontact imaging method based on the radiation force of ultrasound. We extend this technique for tissue characterization of arterial tubes by vibration techniques. The arterial tube can be excited remotely by ultrasound at its resonant frequencies where the vibration and acoustic emission of the tube can be measurable. From these resonant frequencies, the material properties of the arterial tube can be found. A theory for a tube with tissue is formulated using first-order shear deformation theory to include the effects of transverse shear deformation and rotary inertia. A wave propagation approach is applied for easy handling of the boundary conditions. Experimental studies were carried out on a silicone tube embedded in a cylindrical gel phantom. A confocal transducer is used to produce the radiation force of ultrasound for exciting the tube–phantom structure. The vibration of the tube and the phantom are measured with a laser vibrometry system. The fundamental mode of a tube–phantom structure is well excited by the radiation force of ultrasound, and was measured to be 81.8 Hz, which is close to the theoretical prediction of 83.3 Hz. Both excitation and measurement are remote and noncontact, important attributes for future study of arteries.

11:25

3aBB9. A new imaging technique based on resonance for arterial vessels. Xiaoming Zhang, Mostafa Fatemi, and James F. Greenleaf (Ultrasound Res. Lab., Mayo Clinic, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

Vibro-acoustography is a new noncontact imaging method based on the radiation force of ultrasound. We extend this technique for imaging of arterial vessels based on vibration resonance. The arterial vessel is excited remotely by ultrasound at a resonant frequency, at which the vibration of the vessel as well as its transmission to the body surface are large enough to be measured. By scanning the ultrasound beam across the vessel plane and measuring the vibration at one single point on the body or vessel surface, an image of the interior artery can be mapped. Theory is developed that predicts the measured velocity is proportional to the value of the mode shape at resonance. Experimental studies were carried out on a silicone tube embedded in a cylindrical gel phantom of large radius, which simulates a large artery and the surrounding body. The fundamental frequency was measured at which the ultrasound transducer scanned across the tube plane with velocity measurement at one single point on the tube or on the phantom by laser. The images obtained show clearly the interior tube and the modal shape of the tube. The present technique offers a new imaging method for arterial vessels.

11:40

3aBB10. Computerized analysis of bowel sounds in normal and small bowel obstructed rats. Richard Sandler, Hansen Mansy, Michael Uhing, Peter Meyer, and Robert Kimura (Biomed. Acoust. Res. Group, Rush Medical College, 1725 W. Harrison St., Ste. 946, Chicago, IL 60612, rsandler@rush.edu)

Small bowel obstruction (SBO) is a common surgical emergency which may be mimicked by ileus or other nonsurgical conditions. The aims of this work is to delineate gastrointestinal sound (GIS) correlates in a rat model. Seven rats were studied in paired SBO and control states. Computerized analysis of GIS was performed under continuous IV sedation. After adaptive filtering, every GIS event was isolated and analyzed for duration and dominant frequency. It was found that long duration sounds (greater than 100 ms) occurred in each of the obstructed, but in none of the nonobstructed cases ($p=0.02$). The overall mean event duration and dominant frequency in SBO compared to control states was both significantly longer and lower (22.42.6 vs 7.0 ± 2.6 ms, $p=0.001$ for duration; and 296 ± 34 vs 427 ± 33, $p=0.001$ for frequency). Besides these mean differences, there was also a clear evolution with time in GIS characteristics, with lengthening of the duration ($+0.56$ ms/min, $p=0.001$) and lowering of the dominant frequency (-3.3 Hz/min, $p=0.01$). It is concluded that GIS analysis may prove useful in the noninvasive, rapid, and accurate diagnosis of SBO.

11:55

3aBB11. Response of a viscoelastic halfspace to subsurface distributed acoustic sources with application to medical diagnosis. Thomas J. Royston, Yigit Yazicioglu, and Francis Loth (Univ. of Illinois at Chicago, 842 W. Taylor St., MC 251, Chicago, IL 60607, troyston@uic.edu)

The response within and at the surface of an isotropic viscoelastic medium to subsurface distributed low audible frequency acoustic sources is considered. Spherically and cylindrically distributed sources are approximated as arrays of infinitesimal point sources. Analytical approximations for the acoustic field radiating from these sources are then obtained as a summation of tractable point source expressions. These theoretical approximations are compared to computational finite element predictions and experimental studies in selected cases. The objective is to better understand low audible frequency sound propagation in soft biological tissue caused by subsurface sources. Distributed acoustic sources could represent vibratory motion of the vascular wall caused by turbulent blood flow past a constriction (stenosis). Additionally focused vibratory stimulation using a dynamic radiation force caused by interfering ultrasound beams effectively creates a distributed subsurface acoustic source. A dynamic radiation force has been investigated as a means of probing subsurface tissue anomalies, including calcified vascular plaque and tumorous growths. In these cases, there is an interest in relating acoustic measurements at the skin surface and within the medium to the underlying flow/constriction environment or tissue anomaly. [Research supported by NIH NCRR 14250 and Whitaker Foundation BME RG 01-0198.]

Session 3aED

Education in Acoustics, Physical Acoustics and Noise: Demos for all Ages—2003

James M. Sabatier, Cochair
National Center for Physical Acoustics, University of Mississippi, Coliseum Drive, University, Mississippi 38677

Murray S. Korman, Cochair
Department of Physics, United States Naval Academy, Annapolis, Maryland 21402

Matthew E. Poese, Master of Ceremonies
Graduate Program in Acoustics, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16801

The Committee on Education in Acoustics is proud to present a showcase of demonstrations that we think will inspire and captivate a diversified audience as well as motivate further inquiries in the science of sound and associated areas. Please note that presentation times vary. The demonstrations are listed by presenter in alphabetical order but not necessarily in the order of presentation:

Invited Presentations

Modes in the pressure plate of a landmine and their changes with increasing load. W. C. Kirkpatrick Alberts II (Univ. of Mississippi, University, MS 38677)

A laser Doppler vibrometer will be used to measure the vibration of the pressure plate and its output will be shown on a signal analyzer. A load will be added to the pressure plate of the landmine which will demonstrate how the modes change with increasing load.

Spinning cups with sound. Anthony A. Atchley (Graduate Program in Acoustics, Pennsylvania State Univ., 217 Applied Science Bldg., University Park, PA 16802)

After creating a high amplitude two-dimensional standing wave field, first one then four upside-down styrofoam coffee cups can be made to spin individually about a balance point. By changing the relative phases of the field, the cups can be made to change their rotational direction.

Deflection of light by light, Cartesian diver and Super rebound. Lawrence A. Crum (Center for Industrial and Medical Ultrasound, Applied Physics Lab., Univ. of Washington, 1013 NE 40th Street, Seattle, WA 98105)

Three unique demos: 1. ''Deflection of Light by Light'' (actually a thermal gradient), and its relation to bending of sound waves in the ocean, 2. ''Cartesian Diver,'' and its relation to some traveling gliders that APL is developing using the same principle, and 3. ''Super Rebound,'' an interesting phenomenon that occurs when two elastic objects are combined.

A capacitive wave height detector. Jessica Drewery (National Center for Physical Acoustics, 1 Coliseum Dr., University, MS 38677)

The capacitive wave height detector converts low frequency fluid height into a voltage that can be seen on the oscilloscope. From there, you can see the relationship between frequency and wavelength as well as dispersion effects.

Acoustic levitation of droplets and styrofoam pellets in air. E. Carr Everbach (Engineering Dept., Swarthmore College, 500 College Ave., Swarthmore, PA 19081-1397)

Based on a demonstration popularized by Bob Apfel, a 22.4 kHz ultrasonic emulsifier, often used for sonification in biology labs, is used to create an intense standing wave field in air. Small styrofoam pellets from a bean-bag chair and liquid droplets can be levitated and manipulated by audience members with a piece of window screen or hookup wire.

Chladni patterns: Observing normal modes in two dimensional structures. Uwe J. Hansen (Dept. of Physics, Indiana State Univ., Terre Haute, IN 47803)

Sand accumulates along the nodal lines on a plate vibrating at one of the frequencies associated with a normal mode of vibration, thus providing a visual representation of the normal mode pattern. The phase difference of vibrations in adjacent sections separated by a nodal line is observed in the Lissajous figure on an oscilloscope screen when a monitoring microphone crosses a nodal line.

Phase locking. Robert M. Keolian (Applied Research Lab., Pennsylvania State Univ., University Park, PA 16802)

An organ pipe emits a tone whose pitch is normally determined by the pipe's length and the speed of sound. But the organ pipe can change its pitch to match that of another sound source of nearly the same pitch if the two sources are placed close enough together, locking the phases of the pressure swings of the two sounds together and creating one unified sound.

Acoustics at the breakfast table. Andrew Morrison (Physics Dept., Northern Illinois Univ., DeKalb, IL 60115)

The breakfast table is an excellent place to observe some interesting acoustical effects. An empty coffee cup, like an ancient Chinese two-tone bell, emits two distinctly different tones depending upon where it is tapped. When it is filled with hot water and some instant coffee is added, however, a whole new set of sounds are heard when the cup is tapped.

Effects of rooms on voice and music, room mode simulator. Ralph T. Muehleisen (Civil, Environmental and Architectural Engineering, Univ. of Colorado, 428 UCB, Boulder, CO 80309-0428)

Animations and auralizations developed using MATLAB, CATT Acoustics, and Cooledit are used to enhance the understanding of the effects of (1) room architecture on voice and music, and (2) let students simulate measurements of standing waves in a room.

Acoustic log starter. Matthew E. Poese (Graduate Program in Acoustics, Pennsylvania State Univ., P.O. Box 30, State College, PA 16804)

Like a fire starter for a fireplace, a steel tube with a line of small holes drilled along the top is filled with propane and the gas jets are set afire. When the tube is insonified by a loudspeaker placed at one end, the flames above the holes are seen to vary in height along the tube due to second order acoustic effects in the gas contained in the tube.

Demonstration of nonlinear acoustic detection of buried landmines. Waini Karen Tai and Murray S. Korman (Dept. of Physics, U. S. Naval Academy, Annapolis, MD 21402) and James M. Sabatier (Univ. of Mississippi, University, MS 38677)

Airborne sound at two primary frequencies f1 and f2 are chosen at plus and minus 5 Hz from the natural resonance frequency of an inert model landmine that is buried a few inches deep in a sifted soil. A geophone located on the soil surface (over the mine) detects a rich spectrum of nonlinearly generated tones and only very weak nonlinear signals ''off the mine.''

Change of sound speed with temperature. Edward J. Tucholski (Dept. of Physics, U. S. Naval Academy, Annapolis, MD 21402)

The effect of changing the temperature of the medium on the speed of a sound pulse is investigated and displayed.

Rijke tube and singing flame. Ray Scott Wakeland (Graduate Program in Acoustics, Pennsylvania State Univ., P.O. Box 30, State College, PA 16804)

Two unique demonstrations: 1. ''Rijke Tube,'' a tube having a short stack of screens across its diameter, located between a pressure anti-node and node, is made to resonate when the screens are heated. These oscillations require a steady air current through the tube (provided in this case by natural convection) and continue until the screens cool, 2. ''Singing Flame,'' an empty tube can be made to resonate at its fundamental frequency when suspended over a flame from a Bunsen burner.

WEDNESDAY MORNING, 30 APRIL 2003 ROOMS 110/111, 8:30 TO 11:40 A.M.

Session 3aMU

Musical Acoustics and Speech Communication: Singing Voice Acoustics

Thomas Cleveland, Chair

Department of Otolaryngology, Vanderbilt University, 1161 21st Avenue Northeast, Nashville, Tennessee 37232

Chair's Introduction—8:30

Invited Papers

8:35

3aMU1. Can listeners hear who is singing? A comparison of three-note and six-note discrimination tasks. Molly Erickson (Dept. of Audiol. and Speech Pathol., Univ. of Tennessee, Knoxville, TN 37996) and Susan Perry (Univ. of Tennessee, Knoxville, TN 37996)

Timbre is typically investigated as a perceptual attribute that differentiates a sound source at one pitch and loudness; however, singers are believed to have one timbre and singers with similar timbres are thought to comprise a voice category. Yet, by the technical definition, each singer would have a set of timbres across pitch, vowel, and loudness combinations, so that each singer or singing voice category might have a timbre template which allows listeners to recognize one singer or category of singer. This paper investigated the ability of listeners to identify which pitch in an ascending or descending sequence of three or six stimuli was sung by a different singer within and across voice categories. For three-note sequences, the task was difficult and listeners chose the most dissimilarly pitched stimulus as coming from the oddball singer. For six-note sequences, the detection of the oddball singer was much improved, with cross-category comparisons being the easiest. These results support the idea that timbre should be understood as a transformation that connects the different sounds of one source, that a ''rich'' set of exemplars is necessary to discover the trajectory, and that singers of the same category have similar timbre transformations.

8:55

3aMU2. Analysis of levels of support and resonance demonstrated by an elite singing teacher. Ronald C. Scherer, Nandhakumar Radhakrishnan (Dept. of Commun. Disord., Bowling Green State Univ., 200 Health Ctr., Bowling Green, OH 43403), and Andreas Poulimenos (Indiana Univ., Bloomington, IN)

This was a study of levels of singing expertise demonstrated by an elite operatic singer and teacher. This approach may prove advantageous because the teacher demonstrates what he thinks is important, not what the nonsinging scientist thinks should be important. Two pedagogical sequences were studied: (1) the location of support—glottis (poor), chest (better), abdomen (best); (2) locations of resonance—hard palate/straight tone (poor), mouth (better), sinus/head (best). Measures were obtained for a single frequency (196 Hz), the vowel /ae/, and for mezzo-forte loudness using the /pae pae pae/ technique. Sequence differences: The support sequence was characterized by formant frequency lowering suggestive of vocal tract lengthening. The resonance sequence was characterized by flow (AC, mean flow) and abduction increases. Sequence similarities: The best locations had the widest $F2$ bandwidths. The better and best locations had the largest dB difference between $F2$ and $F3$. Although acoustic power increased through the sequences, the acoustic efficiency was not a discriminating factor. Open and speed quotients were not differentiating. The flow resistance was highest and aerodynamic power the lowest for the first of each sequence. Combined data: The maximum flow declination rate correlated highly with the AC flow ($r=-0.92$) and SPL ($r=0.901$).

9:15

3aMU3. Sex and the singer: Gender categorization aspects of singing voice. Sten Ternström (Dept. of Speech, Music and Hearing, Kungliga Tekniska Högskolan, S-100 44 Stockholm, Sweden, sten@speech.kth.se)

The singing voice exhibits many systematic differences by gender and age. The physiological differences between the voice organs of males, females, and children are well known and give rise to several acoustic differences, including acoustic power, pitch range, and spectral distribution. Vocal artists often strive to widen their range of expression, and it is not uncommon for males to sing in a femalelike register, as in counter tenors and in some pop/rock genres. The opposite, however, is quite rare. While ambiguous or contradictory gender in speech is usually a social disadvantage, in singing it can be a desired effect. The physical differences in singing voice production between males and females are reviewed in detail. Some interesting borderline cases are examined from an acoustic standpoint.

9:35

3aMU4. Simulation of singing qualities governed by lower vocal tract adjustments. Ingo R. Titze (Dept. of Speech Pathol. & Audiol., Univ. of Iowa & Natl. Ctr. for Voice & Speech, Denver Ctr. of Performing Arts, 1245 Champa St., Denver, CO 80204, ingo-titze@dcpa.org)

In previous meetings, voice qualities such as pressed, ring, yawn, and twang were discussed in a speech context. It was shown that these qualities have unique spectral characteristics brought about by combinations of glottal and lower vocal tract adjustments (the epilarynx tube and the pharynx). Yawn has a wide glottis, a wide epilarynx tube, and a wide pharynx. On the contrary, twang has a general narrowing of all these airway sections. Ring has a wide pharynx and a relatively narrow epilarynx tube. A pressed voice is primary laryngeal, with a narrowed glottis. In this presentation, similar adjustments are made for singing with a voice simulator that controls vocal tract area functions and glottal flow pulses by rules. Results suggest that various singing styles, such as country-western, opera, or pop, may in part be characterized by these unique combinations of source and filter adjustments.

9:55–10:10 Break

10:10

3aMU5. The singing voice and country music. Wendy D. LeBorgne (The Blaine Block Inst. for Voice Anal. and Rehabilitation, 369 W. First St., Ste. 408, Dayton, OH 45402, wlivar@aol.com)

Preliminary acoustic measures on the Broadway Belt voice suggest uniqueness in this type of vocal production. This study objectively compared the acoustic production of the Broadway Belt voice in four elite and four average belters. Three casting directors evaluated the vocal quality of 20 musical theater majors proficient in the singing style referred to as belting. Each belter sang two specified vocalizes as well as six short excerpts from the belting repertoire. The raters judged the belters on a set of seven perceptual parameters (loudness, vibrato, ring, timbre, focus, nasality, and registration breaks) and reported an overall score. Initially, Pearson product-moment correlation coefficients were calculated and reported for perceived loudness, vibrato, ring, timbre, focus, and nasality for the elite and average groups. Then, significant acoustic results related to vocal intensity, amplitude and magnitude of vibrato, increased spectral energy in the expected Singer's Formant area, and trends in $F1$–$F2$ characteristics were assessed. Overall patterns of these results suggest the elite belters maintained a greater magnitude of vocal vibrato, a brighter vocal quality on some vowels, and different harmonic−−formant relationships than average belters. Specific relevant data related to these acoustical events will be the focus of this presentation.

10:30

3aMU6. Acoustic and aerodynamic characteristics of Country-Western, Operatic and Broadway singing styles compared to speech. Robert E. Stone, Jr. (Retired, Dept. of Otolaryngol., Vanderbilt Univ. Medical Ctr., 5921 Woodland Hills Dr., Nashville, TN 37211, restone@comcast.net), Thomas F. Cleveland (Vanderbilt Univ. Medical Ctr., Nashville, TN 37212), and P. Johan Sundberg (Voice Res. Ctr., Stockholm, Sweden)

Acoustic and aerodynamic measures were used to objectively describe characteristics of Country-Western (C-W) singing in a group of six premier performers in a series of studies and of operatic and Broadway singing in a female subject with professional experience in both styles of singing. For comparison purposes the same measures also were applied to individuals while speaking the same material as sung. Changes in pitch and vocal loudness were investigated for various dependent variables, including subglottal pressure, closed quotient, glottal leakage, H1–H2 difference [the level difference between the two lowest partials of the source

spectrum and glottal compliance (the ratio between the air volume displaced in a glottal pulse and the subglottal pressure)], formant frequencies, long-term-average spectrum and vibrato characteristics (in operatic versus Broadway singing). Data from C-W singers suggest they use higher sub-glottal pressures in singing than in speaking. Changes in vocal intensity for doubling sub-glottal pressure is less than reported for classical singers. Several measures were similar for both speaking and C-W singing. Whereas results provide objective specification of differences between operatic and Broadway styles of singing, the latter seems similar to features of conversational speaking style.

Contributed Papers

10:50

3aMU7. Analysis and enhancement of country singing. Matthew Lee and Mark J. T. Smith (Ctr. for Signal and Image Processing, School of Elec. and Computer Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0250)

The study of human singing has focused extensively on the analysis of voice characteristics. At the same time, a substantial body of work has been under study aimed at modeling and synthesizing the human voice. The work on which we report brings together some key analysis and synthesis principles to create a new model for digitally improving the perceived quality of an average singing voice. The model presented employs an analysis-by-synthesis overlap-add (ABS-OLA) sinusoidal model, which in the past has been used for the analysis and synthesis of speech, in combination with a spectral model of the vocal tract. The ABS-OLA sinusoidal model for speech has been shown to be a flexible, accurate, and computationally efficient representation capable of producing a natural-sounding singing voice [E. B. George and M. J. T. Smith, Trans. Speech Audio Processing **5**, 389–406 (1997)]. A spectral model infused in the ABS-OLA uses Generalized Gaussian functions to provide a simple framework which enables the precise modification of spectral characteristics while maintaining the quality and naturalness of the original voice. Furthermore, it is shown that the parameters of the new ABS-OLA can accommodate pitch corrections and vocal quality enhancements while preserving naturalness and singer identity. Examples of enhanced country singing will be presented.

11:05

3aMU8. Comparison of hearing and voicing ranges in singing. Eric J. Hunter and Ingo R. Titze (Dept. of Speech Pathol. and Audiol., Natl. Ctr. for Voice and Speech, The Univ. of Iowa, Iowa City, IA 52242)

The spectral and dynamic ranges of the human voice of professional and nonprofessional vocalists were compared to the auditory hearing and feeling thresholds at a distance of one meter. In order to compare these, an analysis was done in true dB SPL, not just relative dB as is usually done in speech analysis. The methodology of converting the recorded acoustic signal to absolute pressure units was described. The human voice range of a professional vocalist appeared to match the dynamic range of the auditory system at some frequencies. In particular, it was demonstrated that professional vocalists were able to make use of the most sensitive part of the hearing thresholds (around 4 kHz) through the use of a learned vocal ring or singer's formant. [Work sponsored by NIDCD.]

11:20–11:40

Panel Discussion

WEDNESDAY MORNING, 30 APRIL 2003 ROOM 102, 8:30 TO 10:00 A.M.

Session 3aNSa

Noise: Flow Noise

Courtney B. Burroughs, Chair
Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804-0030

Contributed Papers

8:30

3aNSa1. Prediction of the interior pressure oscillations included by flow over a cavity. Jin-Seok Hong, Jong Beom Park, and Luc Mongeau (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907-1077, hong14@ecn.purdue.edu)

The flow-excited resonance of an open cavity exposed to a grazing flow was investigated experimentally. Flow visualization provided information about the position and the strength of the vortices formed over the orifice. A model based on vortex sound theory was used to predict the frequency and the relative magnitude of the self-sustained interior pressure oscillations. Values for the vortex circulation strength and the convection speed were estimated using a least mean square method. The analytical predictions were compared with experimental values and are in reasonably good agreement with the experimental data. The influence of spoilers and other control devices on the excitation mechanism was investigated.

8:45

3aNSa2. Fan broadband noise from the ingestion of inhomogeneous turbulence. D. A. Lynch, W. K. Blake (NSWC, Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817), and R. Martinez (Anteon Corp./CAA, Cambridge, MA 02140)

Calculations of broadband noise from pre- and post-swirl turbomachines are compared to recent wind-tunnel data. The inflow cases addressed are: (1) grid-generated turbulence for which near-homogeneous conditions prevailed (for the pre-swirl case); and (2) sharply inhomogeneous turbulence encapsulated within the wakes of the rotating upstream blades (for the post-swirl case). Computed results help explain the degree to which strip aerodynamics succeeds in reproducing the noise data, particularly upon comparison to more rigorous estimates that account for spanwise wave numbers in the flow-to-lift transfer function. The paper also reports on the merit of basing the calculation of broadband noise on a circumferentially averaged inflow for case (2), whose turbulence is

thereby rendered artificially homogeneous. The presentation ends with a discussion of the experimental and numerical means whereby the size of the predominant integral scale of the wake-encapsulated turbulent flow was determined, and the extent to which the two sets of values agreed.

9:00

3aNSa3. Acoustical analysis and modeling of reciprocating compressors, noise produced by gas pulsation, using four-pole method. I. Ali T. Herfat and Robert V. Seel (Emerson-Copeland Corp., 1675 W. Campbell Rd., Sidney, OH 45365)

Presented in Paper I are the fluid-structure interactions, structural dynamics, and thermodynamic analyses of reciprocating compressors (such as air conditioning and refrigeration reciprocating compressors). The compressor performance can be analyzed using the follows criteria: (1) thermodynamic model of the cylinder process, using the polytropic process model for thermodynamic model of cylinder; (2) suction and discharge valves dynamics analysis and modeling; (3) the valves modeling and the cylinder volume calculation; (4) Effective flow area and effective force area models.

9:15

3aNSa4. Sources of flow noise due to time-varying tip relief. R. Martinez (Cambridge Acoust. Assoc./Anteon Corp., 84 Sherman St., 3rd Level, Cambridge, MA 02140, rxmartinez@anteon.com)

This paper begins with a brief review of the conventional cause of shed vorticity in two-dimensional unsteady lifting flows around airfoils. It then poses the following three-dimensional theoretical problem in support of experimental work recently conducted at Penn State/ARL: Take a wing tip of arbitrary three-dimensional shape and place a wall near it such that its plane is normal to that of flight. Let the wall be sinusoidally corrugated in the direction of the flow and be convected by it, so that an observer stationed on the wing's tip experiences time-varying influences determined by the wall's passing serpentine pattern. Our presentation will describe a linearizing perturbation analysis whereby the ratio of the wall's sinusoidal amplitude to its wavelength becomes a small parameter. The zeroth-order problem becomes then one of steady three-dimensional aerodynamics barring the presence of external fluctuations. The first-order problem is fundamentally unsteady and driven, as usual, by the zeroth-order one. An unconventional source of flow noise emerges: the ultimately unacceptable axial changes in the strength of trailing vortices, as brought about by their irregular images behind the corrugated wall, lead to new neutralizing shed vortices and harmonic lift.

9:30

3aNSa5. Acoustical analysis and modeling of reciprocating compressors, noise produced by gas pulsation, using four-pole method. II. Ali T. Herfat (Emerson-Copeland Corp., 1675 W. Campbell Rd., Sidney, OH 45365) and Robert V. Seel (Emerson-Copeland Corp., Sidney, OH 45365)

Presented in Paper II is the noise analysis of reciprocating compressors (such as air conditioning and refrigeration reciprocating compressors) using the four-pole method. The gas pulsation noise inside compressor head cavities, mufflers, and through-valves can be analyzed by applying the FPM. This method formulates the characteristics of acoustic elements by establishing a relationship between their input and output gas pressures and volume flow rates. When the acoustic elements in the system (compressor) are connected at points between them, the FPM allows an easy assembly of element equations to obtain system acoustical model.

9:45

3aNSa6. Sound quality metric development for the Air Conditioning and Refrigeration Institute. Kathleen K. Hodgdon and Russell C. Burkhardt (Appl. Res. Lab., Penn State, University, Park, PA 16802)

The current method of assessing acoustic signatures from residential air conditioning units is defined in the Air Conditioning and Refrigeration Institute (ARI 270) 1995 Standard for the Sound Rating of Outdoor Unitary Equipment. This research project was designed to assess the efficacy of the metric to predict consumer preference. The original metric and modified versions of that metric were evaluated as tools for use in today's market. The ARI 270 metric was implemented in software with additional features and the flexibility to modify the features applied. Numerous product acoustic signatures were analyzed and compared using the original metric and various modifications of that metric. As a result of that analysis a set of synthesized signatures was generated that targets problem areas for an application of the metric to typical signature configurations. A subjective jury evaluation was conducted to establish the consumer preference for those synthesized signatures. A statistical correlation of the various configurations was conducted to assess the degree of the relationship between the subjective preferences and the various metric calculations. Recommendations were made for modifications to improve the current metric's ability to predict a subjective preference. [Programming completed by Jonathan Peters (ARL/PSU). Research supported by ARI.]

WEDNESDAY MORNING, 30 APRIL 2003 ROOM 102, 10:15 TO 11:30 A.M.

Session 3aNSb

Noise: Transportation and Community Noise

Bennett M. Brooks, Chair
Brooks Acoustics Corporation, 27 Hartford Turnpike, Vernon, Connecticut 06066

Contributed Papers

10:15

3aNSb1. Aircraft system noise prediction: Past, present, and future. Robert A. Golub and Joe W. Posey (Aeroacoustics Branch, NASA Langley Res. Ctr., M.S. 461, Hampton, VA 23681)

Aircraft system noise prediction is necessary to estimate the community noise impact of future aircraft and to estimate the noise impacts of changes in propulsion systems, airframes, or operations of current aircraft. Aircraft system noise is the sum of noise generated by various components of the propulsion system and the various components of the airframe including the landing gear. Predicting noise on the ground from an aircraft flyover requires estimating the noise generated by the many contributing sources during the flyover as the flight conditions change, summing these sources as a function of time, and propagating the resultant combined source through the atmosphere to the observer location. NASA introduced the Aircraft Noise Prediction Program (ANOPP) about 30 years ago and continually upgraded and extended the code prediction capability. The

history of ANOPP will be reviewed along with current efforts to make it more useful as a design tool. A proposed new systems prediction program, AVATAR, will be less empirical and capable of predicting community noise from unconventional aircraft planforms. Innovative/unconventional aircraft configurations will be required to meet aggressive noise goals in the future.

10:30

3aNSb2. Prediction-based aircraft flyover noise synthesis. Stephen A. Rizzi and Brenda M. Sullivan (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA 23681, s.a.rizzi@larc.nasa.gov)

Investigation of community noise impact of aircraft flyover noise through subjective testing requires time histories of the acoustic pressure at one or more listener positions. The use of recorded aircraft flyover noise for this purpose is often problematic. Factors contributing to this include extraneous noise in the recordings (natural or man-made), a finite number of fixed recording positions, and the cost of conducting the flight test. Perhaps the most limiting factor is the inability to examine proposed aircraft, engines, flight procedures, and other conditions or configurations for which recorded data are unavailable. Synthesis of aircraft flyover noise as an alternative to recordings is thus desirable. An approach to the synthesis of broadband and narrowband aircraft component noise is presented based upon output from community noise prediction codes. These codes typically calculate aircraft flyover noise at designated listener positions in the frequency domain. The necessity to convert these to the time domain presents two challenges both relating to the fact that the predictions are, in essence, time averaged. The first of these involves the reconstruction of phase. The second is the synthesis of temporal fluctuations known to exist in real data. A semiempirical approach to calculating these effects is also presented.

10:45

3aNSb3. Characterization of highway traffic noise generated by rigid pavement contraction joints. Lawrin T. Ellis, Christopher Niezrecki (Dept. of Mech. and Aerosp. Eng., Univ. of Florida, P.O. Box 116250, Gainesville, FL 32611), and David Bloomquist (Univ. of Florida, Gainesville, FL 32611)

Contraction joints in rigid (concrete) pavements are required to permit expansion of each monolithic section of roadway. At higher speeds, the major source of highway noise is attributed to vehicle tire/roadway interaction. Current concerns about limiting the impact of highway traffic noise has forced transportation agencies to consider strategies to control noise generated by tire/roadway interaction. Within this work the difference in noise generated by 1/4- vs 3/8-in. joint widths is conducted. The study focuses on passenger vehicles including a sedan and a light duty van/truck. Both vehicle in-cabin and roadside noise levels are measured for vehicle speeds of 50, 60, and 70 miles per hour. For the sedan, the minimum and maximum observed in-cabin differences were determined to be 1.08 and 1.82 dB(A), respectively. Minimum and maximum observed roadside differences are 1.19 and 2.58 dB(A), respectively. Van tests resulted in minimum and maximum observed in-cabin differences of 0.60 and 1.09 dB(A) and minimum and maximum observed roadside differences of 1.05 and 3.18 dB(A), respectively. This paper contains details of reference standards, test methods, and the results obtained.

11:00

3aNSb4. Chicago Transit Authority train passenger environmental noise study. Matt R. McDuffee and Chris Karner (Columbia College Chicago, 600 S. Michigan Ave., Chicago, IL 60605, Attn: Dominique Cheenne, dcheenne@popmail.colum.edu)

The Chicago Transit Authority (CTA) train system is referred to as the ''L'' because most of the track throughout the city is elevated. Passengers riding aboard the ''L'' are often subjected to high levels of noise due to the aging metal girder system that the tracks are perched on, as well as some train cars that are in disrepair. The environmental acoustics class of nine students at Columbia College Chicago decided to quantify exactly how much noise an ''L'' passenger is subjected to. Using a Quest 2900 integrating sound level meter the class split up and took L_{eq} measurements on all of the seven train lines. Each line was tested in both directions of travel twice, with the meter taking samples every 3 s, which added up to a total of approximately 65 000 samples. The data were then averaged and synthesized into a graph using ESRI ArcView software. The graph is actually a map of the CTA ''L'' system that is color coded according to the L_{eq} level that the passengers are subjected to between each station. It was interesting to see the difference in noise levels according to the type of track construction.

11:15

3aNSb5. A new approach to control noise from entertainment facilities: Active control and measurement of amplified community noise. Richard J. Peppin (Scantek, Inc., 7060-L Oakland Mills Rd., Columbia, MD 21046, PeppinR@ScantekInc.com) and Joan Casamajó (DICESVA, Villar 20, Barcelona 08041, Spain)

While traffic noise is perhaps the most pervasive of community noises, much of the contribution now comes from amplified sound: live music, discos, theme parks, and exercise studios. Those producing the sound or music want it loud and those not interested want to be protected against noise. Noise limits at the receiving or producing property line must be met for the minimum community acceptance. However the time-, and perhaps the spatially-, varying sound in entertainment facilities is often constantly modified (and maybe monitored) near the source of the sound. Hence it is hard to relate and to control the sound at the property line. This paper presents a unique noise control device. It is based on the octave band ''transfer function'' between the sound produced in the entertainment area and the noise received at the property line. The overall insulation can be measured and is input to the instrument. When a noise level limit is exceeded at the receiver, due to the amplified interior noise at the facility, the sound output of the device is automatically controlled to reduce the noise. The paper provides details of the design and possible abatement scenarios with examples.

Session 3aPA

Physical Acoustics, Noise and Engineering Acoustics: Wind Noise in Outdoor Measurements— Characterization and Reduction

Richard Raspet, Chair
National Center for Physical Acoustics, University of Mississippi, Coliseum Drive, University, Mississippi 38677

Chair's Introduction—8:25

Invited Papers

8:30

3aPA1. Turbulent pressure statistics in the atmospheric boundary layer from large-eddy simulation. Natasha L. Miles (Dept. of Meteorol., Pennsylvania State Univ., 503 Walker Bldg., University Park, PA 16802), John C. Wyngaard, and Martin J. Otte (Duke Univ., Durham, NC 27708)

Turbulent pressure fluctuations advected past a sensor contribute to wind noise and can thus significantly degrade acoustic signals. In this study, large-eddy simulation is used to calculate the turbulent pressure field in three types of atmospheric boundary layers. A Poisson equation is used to represent turbulent pressure as the sum of mean-shear, turbulence-turbulence, subfilter-scale, Coriolis, and buoyancy parts. At variance-containing scales in the free-convection case, turbulent-turbulent pressure dominates over the entire boundary layer. It dominates also up to midlayer in the forced-convection case; above that mean-shear pressure dominates. In the stable case mean-shear pressure dominates over the entire layer. Part of the inertial subrange of the pressure spectrum is resolved in the forced- and free-convection cases; it is dominated by the turbulence-turbulence pressure and has a three-dimensional spectral constant of 4.0. This agrees well with quasi-Gaussian predictions but is a factor of 2 less than results from direct numerical simulations at moderate Reynolds numbers. Although past measurements of turbulent pressure have been hampered by instrumental problems, such measurements that could be used to determine the inertial subrange pressure spectral constant would be most useful. [Work supported in part by the U.S. Army through the National Center for Physical Acoustics.]

8:50

3aPA2. Recent progress with atmospheric noise-reducing filters. Michael Hedlin (9500 Gilman Dr., La Jolla, CA 92093-0225)

Wind noise is one of the key issues facing the community of acousticians who monitor the atmosphere for long-period signals from natural atmospheric phenomena or from man-made sources, such as nuclear weapons tests. Presently, a network of infrasound observing sites is being constructed to provide global coverage of acoustic sources in the atmosphere. It is well known that spatial filters can be used to attenuate noise due to atmospheric turbulence near the recording point and increase the ratio of signal-to-noise. Rosette spatial filters are currently used at the new infrasound monitoring stations and have been found to provide a significant reduction of noise across the frequency band of interest to the acoustic monitoring community. Research continues into modifications to existing rosette filters that might improve their performance. Additional research is providing insight into entirely different filters, such as wind barriers. Some of the monitoring systems array sites are located on oceanic islands at locations where spatially extensive noise filters will not be practical. This practical limitation fuels our interest in effective, spatially compact filtering methods. This presentation gives an overview of the various filters currently being used and research into improvements for these filters and into entirely different filter designs.

9:10

3aPA3. The use of an infrasound microphone array to study wind noise spectra and correlation. F. Douglas Shields and Carrick Talmadge (Natl. Ctr. for Physical Acoust., Coliseum Dr., University, MS 38677)

A three dimensional array of infrasound sensors of original design has been constructed and used to study wind generated pressure signals in the frequency range from 0.1 to 100 Hz. The ten sensors in each arm of the array are 2 feet apart. An ultrasonic anemometer ten feet off the ground was used to make simultaneous measurements of the three components of the wind velocity. Several sets of data have been taken in open fields with different ground cover. The data have been spectrally analyzed and, over a limited frequency range, the velocity and pressure variations found to obey the 5/3 and 7/3 power law that is expected for the inertial range. A study has also been made of the dependence of the correlation between the pressure signals and the sensor separation. The coherence of the pressure signals indicates that the convection velocity is nearly independent of frequency, and the correlation has an exponentially decaying sinusoidal dependence on the sensor separation. The array has also been used successfully to localize infrasound sources. [Work supported by the U.S. Army Armament Research Development and Engineering Center.]

9:30

3aPA4. Influence of turbulence frequency and flow unsteadiness on noise reduction of windscreens. Z. Charlie Zheng and Ning Zhang (Mech. and Nuclear Eng. Dept., Kansas State Univ., Manhattan, KS 66506)

Windscreens are widely used in outdoor microphone measurement. However, in many of these applications, low-frequency wind noise interferes with the signals and the performance of measurement microphones significantly depends on the correct design of the windscreen that is used to maximize the signal-to-noise ratio of the sensing system. There are two possible noise sources around a windscreened microphone: one is from the turbulence carried in the incoming flow; the other is from the wake vortex shedding due to interaction between the windscreen and the wind. Recently, we investigated the effects of windscreens on low-frequency wind noise reduction using a steady-state computational fluid dynamic model. The justification of the use of a steady-state flow model was based on experimental data by Morgan and Raspet that showed that when the corresponding frequency was below 10 Hz, the wind noise reduction was almost constant. Under this low-frequency assumption, we found that the wind noise reduction increases with decrease of the Reynolds number. We are currently concentrated on higher frequencies where the wind noise reduction is no longer independent of frequencies and unsteady fluid dynamics is required to provide pressure fluctuation information on the windscreen surface.

9:50

3aPA5. Performance comparisons of microphone windscreens. Edward R. Maniet, Jr. (Textron Systems, 201 Lowell St., Rm. 3101, Wilmington, MA 01887, emaniet@systems.textron.com)

The effectiveness of microphone windscreens is evaluated using a unique low-noise wind tunnel facility that enables controlled, repeatable experiments to be performed [E. R. Maniet, Jr., J. Acoust. Soc. Am. **111**, 2373–2374 (2002)]. Wind noise levels are characterized as a function of frequency, mean wind velocity and turbulent velocity component for several standard windscreen types, including porous foam balls and hollow shell windscreens. The experimental results are used to develop a model for wind noise levels based upon Strasbergs [J. Acoust. Soc. Am. **83**, 544–548 (1988)] dimensional scaling analysis. Also presented is a comparison of microphone wind noise levels for omnidirectional and supercardioid microphone elements.

10:10–10:25 Break

10:25

3aPA6. An investigation of outdoor wind noise reduction by spherical windscreens. Jeremy Webster and Richard Raspet (Dept. of Phys. and Astron., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677)

Phelps [RCA Rev. **3**, 203–212 (1938)] proposed that wind noise pressure fluctuations measured at the center of a spherical wind screen are the area averages resultant of the pressure fluctuations at the surface of the screen. If this hypothesis is applied to the steady state pressure distribution around a spherical windscreen, good agreement between data and theory should be achieved for turblence structures which are large compared to the windscreen. In this paper we report on experiments which extend the work done by Morgan (Ph.D. dissertation, University of Mississippi, 1992) on spherical windscreens. In this experiment, probe microphones placed within reticulated foam windscreens were used in conjunction with anemometers placed directly in the airflow near the windscreen to investigate the correlation between incident flow and the resulting pressure measurements throughout the foam.

Contributed Papers

10:45

3aPA7. Effect of atmospheric pressure fluctuations on low frequency and infrasound detection. Carrick L. Talmadge and Doug Shields (Univ. of Mississippi, NCPA, Oxford, MS 38677, clt@olemiss.edu)

A fundamental difficulty in low-frequency and infrasound detection is the so-called "wind noise" problem. At higher (>50 Hz), this wind noise is usually associated with the obstruction formed by the pressure pobe body. In this case, most of this wind noise can be adequately removed by the use of an appropriate wind screen. At lower frequencies (<50 Hz), the magnitude of the intrinsic atmospheric fluctuations appears to be much larger than the flucutations induced by the sensor body. A new large-element (>30 elements) 3-axis pressure sensor array is used to quantify the relative magnitude of the intrinsic and induced pressure fluctuations. Signal processing algorithms are discussed which enchance the signal-to-noise ratio over that obtained using standard beam forming algorithms, by utilizing the localized nature of many of the fluctuations across the array. The effect of the height of the sensor above ground on the magnitude of the observed pressure flucutations is also examined using a 3-axis linear pressure sensor array. Finally, the performance of a single-element porous hose array is compared to that of a large-element array.

11:00

3aPA8. Measuring immeasurable sound pressure levels. Kenneth E. Gilbert, Carrick L. Talmadge (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, Coliseum Dr., University, MS 38677), and Xiao Di (Appl. Res. Lab., Penn State Univ., State College, PA 16804)

The AWSUM(K) processors developed by R. A. Wagstaff make possible the detection of narrowband signals deeply buried in wind noise. However, the signal-to-noise ratio one obtains with the AWSUM(K) processors is not the ratio of signal power to noise power, but a nonphysical ratio that depends on the coherence of the signal and the noise distribution. A recently developed theory, used in conjunction with the AWSUM(K) processors, allows the recovery of a physically meaningful signal-to-noise ratio, even when wind noise prevents a direct measurement of the signal. Thus, it is possible not only to detect a signal buried in wind noise, but also to estimate the physical sound pressure level. Such a capability can be important for identifying a source using, for example, the ratio of harmonic amplitudes produced by a particular engine in a ground vehicle. The theory for recovering the physical sound pressure level will be discussed and applications to field data will be presented.

11:15

3aPA9. Wind noise and the spectrum of atmospheric turbulence pressure fluctuations. D. Keith Wilson (U.S. Army Cold Regions Res. and Eng. Lab., 72 Lyme Rd., Hanover, NH 03755)

Previous research [S. Morgan and R. Raspet, J. Acoust. Soc. Am. **92**, 1180–1183 (1992)] has shown that wind noise is predominantly caused by pressure fluctuations intrinsic to the turbulent atmospheric flow. Therefore it should be possible to predict wind noise from models for turbulent pressure spectra in the atmosphere. Based on simple dimensional analysis and an application of Taylor's hypothesis, the inertial-subrange power spectrum for turbulent pressure fluctuations should be proportional to $f^{-7/3}$, where f is frequency. But more recent atmospheric observations and theoretical arguments [J. D. Albertson, G. G. Katul, M. B. Parlange, and W. E. Eichinger, Phys. Fluids **10**, 1725–1732 (1998)] suggest that the power spectrum actually goes as $f^{-3/2}$. In this paper, it is shown that the $f^{-3/2}$ dependence predicts a much slower decay in wind noise with increasing acoustic frequency than is typically observed. Possible reasons for this discrepancy are discussed.

11:30

3aPA10. A feasibility study of air-coupled ultrasonic vibrometers. Andi Petculescu and James Sabatier (Natl. Ctr. for Physical Acoust., 1 Coliseum Dr., University, MS 38677, apetcule@olemiss.edu)

Several key issues arising when an ultrasonic field is used to probe small surface vibrations are presented. Sum and difference frequencies (sidebands) can be produced by the Doppler shift of the primary probe frequency incurred at the vibrating boundary and by the nonlinear parametric interaction with the radiation from the oscillating surface. Carrier and surface vibration amplitudes, the condition of the oscillating surface, and ambient parameters, influence the received signal. Wind critically affects the reliable operation of air-coupled acoustic sensors. As wind energy increases, the sensor detects a progressively richer turbulent spectrum that drowns the information-bearing sidebands. Experimental data of the scattering of ultrasonic energy by a turbulent environment is shown, aimed at searching for ways to actively recover the vibration sidebands. [Work supported by ONR.]

WEDNESDAY MORNING, 30 APRIL 2003 ROOM 204, 8:00 A.M. TO 12:00 NOON

Session 3aPP

Psychological and Physiological Acoustics and Animal Bioacoustics: Honoring the Contributions of Jozef Zwislocki

Robert D. Frisina, Chair

Otolaryngology Division, University of Rochester Medical Center, 601 Elmwood Avenue, Rochester, New York 14642-8629

Chair's Introduction—8:00

Invited Papers

8:10

3aPP1. Jozef Zwislocki: Impact on models of coding in the auditory nerve. Murray B. Sachs (Dept. of Biomed. Eng., Ross 720, Johns Hopkins Univ., Baltimore, MD 21205, msachs@bme.jhu.edu)

The auditory nerve has long been considered a window on the biophysical mechanisms of cochlear transduction and the most carefully characterized aspect of the responses of single auditory-nerve fibers has been the tuning curve. Perhaps the most intensively studied question in auditory theory is: What is the relationship between the shapes of these tuning curves and basilar membrane displacements? The basilar membrane measurements of Georg von Bekesy stimulated a generation of basilar-membrane modelers, none more notable than Joe Zwislocki, who was awarded the first von Bekesy Medal by the Acoustical Society in 1985. The impact of Zwislocki's basilar membrane models on our understanding of auditory nerve tuning will be reviewed. The properties of auditory-nerve discharge patterns are also shaped by the filtering properties of the hair cell/synapse complex. The major contributions of Joe and his students to our understanding of this filtering through their elegant experimental and modeling studies of adaptation in the auditory nerve will be presented. Throughout his career, Joe Zwislocki has maintained an active interest in loudness summation and his work in relating the input/output characteristics of auditory-nerve fibers to loudness will be highlighted.

8:35

3aPP2. Jozef Zwislocki in the post-von Bekesy era of cochlear physiology. William S. Rhode (Univ. of Wisconsin, 1300 University Ave., Madison, WI 53706, rhode@physiology.wisc.edu)

Professor Joe Zwislocki is a rare individual who has a strong background in mathematics and hydrodynamics that he applied to the physiological characterization of cochlear function. Realizing the preeminent need for more accurate measurements of cochlear mechanical properties, he undertook the study of properties of the tectorial membrane (TM). His measurements are unique in that they remain the only *in vivo* measurements of stiffness of the TM. In 1979, he introduced longitudinal effects into cochlear modeling that today has been shown to result in realistic cochlear mechanical transfer curves. A difference between the filter properties seen in auditory nerve fibers and basilar membrane (BM) mechanics led to a leading theory of separate resonances for organ of Corti and TM. He developed a model of how this would affect the shear motion between the TM and reticular lamina. Additional studies were undertaken that substituted recordings in the cells of Hensen for the very difficult direct observation of BM mechanics that epitomize the ingenuity and resourcefulness of Joe Zwislocki.

9:00

3aPP3. Jozef Zwislocki's integrated approach to psychoacoustics. Rhona P. Hellman (Dept. of Psych., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, hellman@neu.edu)

Early in the psychophysics phase of his illustrious and varied scientific career, Joe Zwislocki recognized that loudness is of key importance for understanding signal processing by the auditory system. To better comprehend just how loudness can be incorporated into basic auditory theory, Joe sought to determine the loudness-intensity relation down to near threshold levels. Together with Hellman, this aim motivated the development of absolute magnitude-scaling procedures. Later, Joe demonstrated that absolute scaling yields results compatible with nonmetric measures of loudness additivity. His search for a comprehensive theory of auditory function led him to deduce that both the observed proportionality between loudness and sound intensity near threshold and the compressive nonlinearity in the loudness function at moderate-to-high levels are generated by the peripheral auditory system. These innovative concepts were incorporated into his theoretical analyses of temporal summation and central masking. They also provided the basis of a loudness model formulated to describe loudness growth in quiet and in noise. Not only are Joe Zwislocki's extraordinary insights compatible with recent loudness measures, his loudness model can be extended successfully to predict the growth of loudness characteristic of cochlear-impaired hearing. [Work supported by NEDO, Japan.]

9:25

3aPP4. Understanding cochlear function through auditory-nerve activity: A Zwislocki perspective. Richard A. Schmiedt (Dept. of Otolaryngol.–Head and Neck Surgery, P.O. Box 250550, Medical Univ. of South Carolina, Charleston, SC 29425, schmiera@musc.edu)

Dr. Joe Zwislocki served as my dissertation advisor during those tumultuous years in cochlear physiology when our understanding of outer hair cell (OHC) function was evolving from that of a sensory cell to that of a mechanical amplifier. Spoendlin had recently demonstrated that 90%–95% of auditory-nerve afferents originated from inner hair cells (IHCs), but the characteristics of IHC receptor potentials remained an enigma. Otoacoustic emissions and OHC electromotility were terms yet to be defined. Theories relating auditory-nerve activity to basilar-membrane mechanics included concepts of second filters, basilar-membrane nonlinearities, and phase opposition. It was a fertile time for theories and experiments attempting to describe a black-box system that did not yield its mysteries easily. Around 1977, IHC receptor potentials were found to be as sharply tuned as auditory-nerve responses, and the era of cochlear micromechanics began. Joe Zwislocki, as usual, has played a primary role in defining this new era, utilizing the relationships between the OHC stereocilia and the tectorial membrane as his modeling clay.

9:50

3aPP5. Adapting to Jozef Zwislocki. Robert L. Smith (Depts. of Bioengineering & Neurosci. and the Inst. of Sensory Res., Syracuse Univ., 621 Skytop Rd., Syracuse, NY 13244, bob_smith@isr.syr.edu)

Dr. Joe Zwislocki has been a source of excellent advice throughout my career. However, two very important things he told me were wrong as will be disclosed in the presentation. Nevertheless, among the many fundamental things I learned from him was how one must attempt to take the data, and turn it over and over again until the hidden patterns and quantitative inter-relationships emerge. It was because of Joe that I first began to study auditory-nerve adaptation. His challenge to me was to find a model of adaptation that was not only functionally useful but also physiologically realizable, and that quest is still continuing, on my part and that of many others in our field. When Joe suggested auditory-nerve adaptation as a possible source of psychophysical overshoot, our analysis showed that could be true, but only for an on-frequency overshoot of several dB. Consequently the sources of the large off-frequency overshoot remain enigmatic. New results showing overshoot in cochlear implant listeners will be presented and some implications for acoustic overshoot discussed. However, my overall conclusion is that I still have not adapted to Joe Zwislocki, and he continues to be a significant ongoing source of new insights and analysis.

10:15–10:30 Break

10:30

3aPP6. Understanding loudness: Testing physiological models of intensity discrimination and masking for their consistency with loudness functions. Laurel H. Carney (Depts. of Bioengineering & Neurosci. and the Inst. of Sensory Res., Syracuse Univ., 621 Skytop Rd., Syracuse, NY 13244, laurel_carney@isr.syr.edu)

Joe Zwislocki continues to contribute actively to ongoing scientific discourse at the Institute for Sensory Research (ISR). Naturally, these discussions include, amongst other topics, the relationship between loudness and intensity discrimination. Being in the ISR environment has provided the motivation to test models for intensity discrimination and masking in the context of loudness. This problem is timely in that it relates to Joe's current efforts to assemble a written summary of his contributions to the study of loudness. In this presentation, the ability of recent neural models for intensity discrimination and masking to explain some basic aspects of the loudness functions reported by Joe Zwislocki and his colleagues will be explored. [Work supported by NIDCD-NIH.]

10:55

3aPP7. Jozef Zwislocki's contribution to the understanding of cutaneous sensation. Stanley J. Bolanowski (Inst. for Sensory Res., Dept. of Bioengineering & Neurosci., 621 Skytop Rd., Syracuse Univ., Syracuse, NY 13244, Sandy_Bolanowski@isr.syr.edu)

Whereas Professor Zwislocki is well known for his theoretical and experimental activities that discovered many principles about the auditory system as outlined in this special session, his influence on research efforts and contributions to the knowledge base of the cutaneous sensory system has not been as widely appreciated. Philosophically, he believes that all of the sensory systems have common, as well as different capabilities, and it is this philosophy which led him to explore many of the underlying factors behind

somatosensation. This presentation will outline his scientific and philosophical input to the understanding of somatosensation from the level of receptor function to higher cognitive aspects. For example, he has influenced various views regarding tactile psychophysical thresholds and the relationships between sensation magnitude and the Differenz Limen. His theories on temporal summation and thoughts regarding independent tactile channels of communication originating in the periphery and passing on to the central nervous system will also be discussed. Physiologically he was a prominent player in determining transduction mechanisms of one of the prototypical mechanoreceptors found within the skin, the Pacinian corpuscle. Indeed, how somatosensation comes about has progressed greatly from his oftentimes unrealized influence.

11:20

3aPP8. The "Zwislocki effect" in my work with cochlear implants. Monita Chatterjee (House Ear Inst., Dept. of Auditory Implants & Percept., 2100 W. Third St., Los Angeles, CA 90057, monita@hei.org)

Joe Zwislocki's uniqueness as a scientist lies in his ability to transcend methodology, sensory modality, or system—he has spanned the realms of mathematics, engineering, physics, physiology, and psychophysics to study hearing, touch, and vision. As a graduate student, I was amazed and inspired by his unerring ability to find unifying patterns underlying seemingly different phenomena. I became interested in cochlear implants while a graduate student in Joe's lab. Cochlear implants present a unique opportunity to the sensory scientist: by bypassing the complex frequency analysis of the cochlea and directly stimulating the auditory nerve, they allow us to separate aspects of temporal and spectral (tonotopic) processing by higher centers of the auditory system. In a series of recent experiments, we have exploited this feature of cochlear implants to better understand the processing of complex signals by the auditory system. In this presentation, I will review these and other experiments on loudness, masking, and temporal processing in electrically stimulated hearing. Although Joe has not (as yet) worked on cochlear implants, his approach to sensory systems is undoubtedly the strongest scientific influence (the "Zwislocki effect") in my work in this area.

Contributed Paper

11:45

3aPP9. A comparison of Levitt and Zwislocki decision rules for use with forced-choice adaptive procedures. Robert S. Schlauch and Edward J. Carney (Dept. of Commun. Disord., 164 Pillsbury Dr. SE, Minneapolis, MN 55455, Schla001@umn.edu)

Forced-choice adaptive procedures enjoy widespread use for the measurement of detection and discrimination thresholds. Zwislocki *et al.* [J. Acoust. Soc. Am. **30**, 254–262 (1958)] proposed an adaptive procedure with a decision rule that targets 75% correct several years before these procedures gained acceptance in psychophysics, and even today little is known about the statistical properties of this decision rule. This paper evaluates, using computer simulations, the bias and efficiency of Zwislocki's proposed decision rule in 2-alternative forced-choice (AFC) and 3-AFC procedures under conditions of full attention and inattenion. The results for this decision rule compare favorably with two popular rules proposed by Levitt [J. Acoust. Soc. Am. **49**, 467–477 (1971)] that target 70.7% correct and 79.4% correct. In summary, the rule that targets 75% correct (Zwislocki's rule) produces less biased threshold estimates than the rule that targets 70.7% correct and it is affected less by inattention than the rule that targets 79.4% correct. This research supports the addition of Zwislocki's rule to the psychophysical toolbox.

WEDNESDAY MORNING, 30 APRIL 2003 ROOMS 105/106, 8:00 TO 9:45 A.M.

Session 3aSAa

Structural Acoustics and Vibration and Signal Processing in Acoustics: Interior Noise in Aircraft and Rocket Fairings

Robert L. Clark, Chair

Department of Mechanical Engineering and Material Science, Duke University, Box 90300, Durham, North Carolina 27708-0300

Chair's Introduction—8:00

Invited Papers

8:05

3aSAa1. Development of a device for reduction of low-frequency sound transmission reduction in small launch vehicles. Steven F. Griffin and Steven A. Lane (Boeing SVS, 4411 The 25 Way NE, Ste. 350, Albuquerque, NM 87109)

Launch acoustic loads have the potential to damage sensitive payloads within a payload fairing, often requiring more structural mass to withstand these loads than would otherwise be necessary to survive launch. A typical payload fairing will exhibit several cavity resonance modes related to its axial length. These modes are excited both by the vibrations transmitted through the structure during launch, and by acoustic ground reflections coupled into the system during the initial launch phase. Conventional acoustic blankets mitigate the acoustic environment within a payload fairing, but are generally only effective above 250 Hz. In this work we

present an Adaptive Vibro-Acoustic Device (AVAD), which is designed to actively and passively absorb acoustic energy in a payload fairing at frequencies below 250 Hz. To date, a prototype has been designed and tested for application to a sounding rocket experiment, Vibro-Acoustic Launch Protection Experiment (VALPE), being conducted by the Air Force Research Laboratory. Prototype test results as well as projected performance in flight will be presented.

8:30

3aSAa2. Limits on the performance of Distributed Vibration Absorbers for the control of broadband disturbances. Marty Johnson, Tony Harris, and Chris Fuller (Vib. and Acoust. Labs., Virginia Tech, VA 24061-0238)

Distributed Vibration Absorbers (DVAs) have been shown to be useful passive control devices for reducing the vibration levels on lightly damped aerospace structures subjected to broadband excitation. These devices work by coupling to the modes of the structure and efficiently adding damping. One advantage of the DVA over the classic vibration absorber (VA) is that it acts over a large area instead of at a single attachment point. This paper investigates how this spatial distribution allows the absorber mass to be re-used multiple times in order to control multiple modes. In order to quantify the benefit of using a DVA this paper compares the mass of the DVA under investigation to the mass of a set of VAs that produce the same vibration reduction. This mass ratio (R) is therefore a measure of the effectiveness of the DVA. The effect of the complexity of the DVA design and the spatial extent of the DVA on R, are both investigated and results are presented. Using a computational model of a cylinder it will be shown that large mass reductions can be achieved using DVAs but that they must act over a large area and are susceptible to variations in structural properties. [Work supported by Boeing.]

8:55

3aSAa3. Active control of payload fairing noise using distributed active vibration absorbers. Arnaud Charpentier (Vibro-Acoust. Sci., Inc., 12555 High Bluff Dr., Ste. 310, San Diego, CA 92130, arnaud@vasci.com), Marty E. Johnson, and Chris R. Fuller (Virginia Tech, Blacksburg, VA 24060)

High sound pressure inside a launch vehicle fairing during lift-off can damage the payload. Interior levels of up to 140 dB between 60 and 250 Hz are mostly due to exhaust plume noise combined with the limited transmission loss of lightweight composite fairings and little acoustic damping in the fairing volume. Past work using passive and hybrid passive/reactive noise control devices has shown that their limitations are mostly due to packaging volume and weight penalty. The objective of this work is to design a lightweight, compact, and low electrical power active noise control system to reduce the fairing interior sound level. Hybrid active/passive actuators such as Smart Foam (Couche and Fuller, *Proceedings of Active 1999*, Ft. Lauderdale, FL, pp. 609–620) and Distributed Active Vibration Absorbers (Marcotte, Fuller, and Johnson, *Proceedings of Active 2002*, ISVR, Southampton, England, pp. 535–546) are optimized for fairing noise control. The latter have been used to increase the transmission loss of the fairing. Active noise control test results on a sub-scale, sandwich composite fairing are presented. The global interior acoustic response due to airborne exterior excitation is minimized using an adaptive multiple-input, multiple-output feedforward controller. [Work supported by the Air Force Research Laboratory, Space Vehicles Directorate (AFRL).]

3a WED. AM

9:20

3aSAa4. The control of rocket fairing interior noise with a networked embedded system. Kenneth D. Frampton (Dept. of Mech. Eng., Vanderbilt Univ., VU Station B 351592, Nashville, TN 37235)

Numerous investigations have been conducted with the purpose of attenuating the acoustic environment within rocket payload fairings. These, to date, theoretical and experimental laboratory studies have demonstrated a great deal of success. However, practical applications to this, and other large-scale noise control problems, have been limited in their success. These limitations are due to nonscalable control systems, weight constraints and complexity. This work seeks to address these limitations by investigating the use of an array of networked embedded processors to control the interior acoustics of a rocket fairing. This networked embedded system consists of numerous computationally elements, paired with appropriate sensors and actuators, that communicate with each other over a wired or wireless network. The goal of the network is to minimize the interior acoustic level while expending a minimum amount of energy. Results from the simulation of such a control system will demonstrate the effectiveness of such an approach. These results will also be compared with those obtained by traditional, centralized control architectures.

Session 3aSAb

Structural Acoustics and Vibration: Computational Methods

Lonny L. Thompson, Chair
Department of Mechanical Engineering, Clemson University, 219 EIB, Box 340921, Clemson, South Carolina 29634-0921

Contributed Papers

10:00

3aSAb1. Adaptive time-discontinuous Galerkin finite element methods for acoustic scattering. Dantong He and Lonny L. Thompson (Dept. of Mech. Eng., Clemson Univ., Clemson, SC 29634-0921)

Comprehensive self-adaptive procedures with efficient sparse multi-level iterative solution algorithms for the time-discontinuous Galerkin space–time finite element method (DGFEM) including high-order accurate nonreflecting boundary conditions (NRBC) are developed for acoustic scattering problems. An *h*-adaptive space–time strategy is employed based on a superconvergent patch recovery (SPR) technique, together with a temporal error estimate. The use of sub-time steps with pre-integrated local space–time elements to efficiently track waves in space–time are also demonstrated. For accurate data transfer (projection) between meshes, new superconvergent interpolation (SI) procedures are developed. Numerical studies of transient acoustic scattering demonstrate the accuracy, reliability and efficiency gained from the adaptive strategy. [Work supported by NSF.]

10:15

3aSAb2. Parallel iterative solution of large-scale acoustic scattering problems using exact non reflecting conditions on distributed memory computer systems. Cristian Ianculescu and Lonny L. Thompson (Dept. of Mech. Eng., Clemson Univ., Clemson, SC 29634-0921)

Parallel iterative methods for fast solution of large-scale acoustic radiation and scattering problems are developed using exact Dirichlet-to-Neumann (DtN) nonreflecting boundaries. For elongated scatterers such as submarines, it is shown that the generalization of the DtN to elliptical/spheroidal artificial boundaries improves significantly the computational efficiency of accurate finite element methods for the solution of acoustic scattering problems. The outer-product structure of the DtN map is exploited as a low-rank update of the system matrix to efficiently compute the matrix-by-vector products found in Krylov subspace based iterative methods. For the complex non-Hermitian matrices resulting from the Helmholtz equation, a distributed-memory parallel BICG-STAB iterative method is used in conjunction with a hybrid parallel SSOR/Jacobi preconditioner. The domain decomposition with interface minimization was performed to ensure optimal inter-processor communication. For the distributed memory architectures tested, including Linux/Intel Beowulf clusters, when implemented as a low-rank update, the nonlocal character of the DtN map shows little impact on the scale up or parallel efficiency compared to approximate local boundary conditions. [Work supported by NSF.]

10:30

3aSAb3. Hybrid NAH formulations for an arbitrary object in a nonfree field. Sean Wu (Dept. of Mech. Eng., Wayne State Univ., Detroit, MI 48202)

Previous nearfield acoustical holography (NAH) formulations based on the Helmholtz integral theory and boundary element method (BEM) are effective for an arbitrary object in a free field, although a true free field is nonexistent. While a modified Helmholtz equation least squares (HELS) method using expansions of both outgoing and incoming spherical waves can provide an approximate solution for such a scenario [Wu and Zhao, J. Acoust. Soc. Am. **111**, 2409 (2002)], the reconstruction accuracy may be unsatisfactory when the source is of arbitrary geometry. In this paper, a hybrid NAH formulation that combines a modified HELS method, the Helmholtz integral theory, BEM, and regularization techniques (e.g., modified Tikhonov regularization, generalized cross validation) is developed for reconstructing acoustic radiation from an arbitrary object in a nonfree field. This hybrid formulation has the advantages of the modified HELS method and BEM-based NAH, and represents a significant improvement over the combined HELS method [Wu and Zhao, J. Acoust. Soc. Am. **112**, 179–188 (2002)]. The input data are collected on a conformal surface at close range, so the evanescent waves can be captured and the reconstruction accuracy can be improved. Moreover, the majority of the input data are calculated but not measured, and the reconstruction efficiency is enhanced. [Work supported by NSF.]

10:45

3aSAb4. Further developments of a high-frequency broadband energy-intensity boundary element method. Jerry W. Rouse and Linda P. Franzoni (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708-0300, jwrouse@me1.egr.duke.edu)

The prediction of the spatial mean square pressure distribution within enclosed high-frequency broadband sound fields is computationally intensive if determined on a frequency-by-frequency basis. Recently an energy-intensity boundary element method has been developed which employs uncorrelated broadband directional energy sources to predict expeditiously such pressure distributions. The source strength and directivity associated with the energy sources, distributed over enclosure boundaries, were determined in an iterative manner. Here, further refinement of the method is presented including the direct determination of source strength and directivity without iteration. Additionally, recent improvements allow for specifying the degree to which the reflected field is modeled as diffuse relative to specular for a given element. Good agreement between the improved energy–intensity boundary element method and experimental measurements and/or exact analytical solutions are shown. [Work sponsored by NSF.]

11:00

3aSAb5. An expanded spherical wave expansion for arbitrary sound fields. Nassif E. Rayess (Univ. of Detroit Mercy, Detroit, MI 48219)

Spherical wave function expansions as means of describing the acoustic field radiated by arbitrarily shaped objects are very convenient and gaining popularity. The HELS method for acoustic holography is one methodology advocating the use of spherical wave functions as a basis for the solution of the inverse acoustic problem. Such methodologies only provide an approximation to the actual sound field and as such suffer from errors, albeit understood to be bounded and manageable in a large number of cases. Based on the sound radiation model of a volume distribution of monopole sources, these approximation errors are found to be due to the presence of monopole sources outside the largest inscribed virtual sphere. Monopole sources outside that virtual sphere are not accounted for in the expansion and thus come out as errors. These errors are greatly reduced if the spherical wave function expansion is modified to account for the largest number of monopole sources in the model. This is accomplished by employing the addition theorem for spherical harmonics. This modifica-

tion complicates the problem mathematically and might require a greater rate of physical sampling, however the reduction in the reconstruction errors might prove beneficial.

11:15

3aSAb6. The use of a waveguide expansion to model landmine detection using acoustic to seismic coupling. Doru Velea (Planning Systems, Inc., 12030 Sunrise Valley Dr., Reston Plaza I, Ste. 400, Reston, VA 20191) and Roger Waxler (Natl. Ctr. for Physical Acoust., The Univ. of Mississippi, 1 Coliseum Dr., University, MS 38655)

The use of a waveguide of sufficiently large radius to simplify the modeling of the infinite space response of a buried landmine to airborne sound has been investigated. It was previously determined that if the ground is modeled as an effective fluid, an efficient and rapidly converging algorithm can be obtained. The effective fluid has been replaced with an elastic solid. For such a model, this technique fails to converge sufficiently rapidly to the infinite space limit. The surface wave supported by the shear sector gets excited by the landmine. The decay of this surface wave with distance is too slow, forcing one to use a waveguide of a prohibitively large radius. In conclusion, this technique at best gives qualitative results. [Work sponsored by the U.S. Army Communications–Electronics Command, Night Vision and Electronics Sensors Directorate.]

11:30

3aSAb7. Acoustical wave propagator technique for time-domain analysis of dynamic stress in a step plate. Shuzhi Peng and Jie Pan (School of Mech. Eng., The Univ. of Western Australia, 35 Stirling Hwy., Crawley, WA 6009, Australia, speng@mech.uwa.edu)

In this paper, we introduce an explicit acoustical wave propagator technique to describe the time-domain evolution of acoustical waves in two-dimensional plates. This technique uses a combined scheme with Chebyshev polynomial expansion and fast Fourier transformation for implementation of the operation of the acoustical wave propagator. We also apply the acoustical wave propagator for studying dynamic stress in a step plate in time-domain.

11:45

3aSAb8. Ecological prognosis near intensive acoustic sources. Stanislav A. Kostarev (Lab. of Acoust. and Vib. Tunnel Assoc., 21 Sadovo-Spasskaya Str., Moscow 107217, Russia), Sergey A. Makhortykh (Russian Acad. of Sci., Pushchino, Moscow reg. 142290, Russia), and Samuil A. Rybak (N. N. Andreev Acoust. Inst., Moscow 117036, Russia)

The problem of a wave field excitation in a ground from a quasi-periodic source, placed on the ground surface or at some depth in soil is investigated. The ecological situation in this case in many respects is determined by quality of the raised vibrations and noise forecast. In the present work the distributed source is modeled by the set of statistically linked compact sources on the surface or in the ground. Changes of parameters of the media along an axis and horizontal heterogeneity of environment are taken into account. Both analytical and numerical approaches are developed. The last are included in software package VibraCalc, allowing to calculate distribution of the elastic waves field in a ground from quasilinear sources. Accurate evaluation of vibration levels in buildings from high intensity under ground sources is fulfilled by modeling of the wave propagation in dissipative inhomogeneous elastic media. The model takes into account both bulk (longitudinal and shear) and surface Rayleigh waves. For the verification of used approach a series of measurements was carried out near the experimental part of monorail road designed in Moscow. Both calculation and measurements results are presented in the paper.

WEDNESDAY MORNING, 30 APRIL 2003 ROOM 206, 8:00 A.M. TO 12:00 NOON

Session 3aSC

Speech Communication: Voices in the Neighborhood: Production, Perception and Anatomy (Poster Session)

Lori L. Holt, Chair
Department of Psychology, Carnegie Mellon University, 5000 Forbes Avenue, Pittsburgh, Pennsylvania 15213

Contributed Papers

All posters will be on display from 8:00 a.m. to 3:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon. To allow for extended viewing time, posters will remain on display until 3:00 p.m.

3aSC1. Gestural overlap of stop-consonant sequences: Evidence from analysis and synthesis. Sherry Zhao and Kenneth N. Stevens (Res. Lab. of Electron. and Dept. of EECS, MIT, Cambridge, MA 02139, szhao@mit.edu)

This study uses an analysis-by-synthesis approach to discover possible principles governing the coordination of oral and laryngeal articulators in the production of English stop-consonant sequences. Individual recordings were made of two male and two female native American-English speakers reading phrases which include voiced and voiceless stop consonants in word-initial (V#CV) and word-final (VC#V) positions, as well as in VC#CV stop–stop consonant sequences. Articulatory timing estimates were made based on analyzing acoustic data including formant movements, closure durations, release bursts, and spectrum shape at low frequencies. Based on the gestural estimates, the same consonant sequences were generated using HLsyn, a quasiarticulatory synthesizer. The synthetic utterances were acoustically and perceptually compared to the actual utterances in order to verify and refine the articulatory timing estimates from which possible principles could be derived. Preliminary results agree with earlier findings of more overlapping of oral gestures in sequences with front-to-back order of place of articulation than those with back-to-front order [Chitoran, Goldstein, and Byrd, Lab. Phonology **7**, 419–448 (2002)]. Furthermore, overlapping of laryngeal gestures is suggested by the smaller acoustical loss at the glottis in vowels after voiced–voiceless sequences than voiceless–voiceless sequences.

3aSC2. Infants in cocktail parties. Rochelle S. Newman (Dept. of Hearing & Speech Sci., Univ. of Maryland, College Park, MD 20742, rnewman@hesp.umd.edu)

Most work on listeners' ability to separate streams of speech has focused on adults. Yet infants also find themselves in noisy environments. In order to learn from their caregivers' speech in these settings, they must first separate it from background noise such as that from television shows and siblings. Previous work has found that 7.5-month-old infants can separate streams of speech when the target voice is more intense than the distractor voice (Newman and Jusczyk, 1996), when the target voice is known to the infant (Barker and Newman, 2000) or when infants are presented with an audiovisual (rather than auditory-only) signal (Hollich, Jusczyk, and Newman, 2001). Unfortunately, the paradigm in these studies can only be used on infants at least 7.5 months of age, limiting the ability to investigate how stream segregation develops over time. The present work uses a new paradigm to explore younger infants' ability to separate streams of speech. Infants aged 4.5 months heard a female talker repeat either their own name or another infants' name, while several other voices spoke fluently in the background. We present data on infants' ability to recognize their own name in this cocktail party situation. [Work supported by NSF and NICHD.]

3aSC3. Cocktail party effect in infants: Visual information and speech segmentation in noise. George Hollich (Dept. of Psych. Sci., Purdue Univ., 703 Third St., West Lafayette, IN 47907-2004, ghollich@purdue.edu)

What are infants abilities to use what they see to segment speech in a noisy environment? Infants often find themselves in situations far louder and more complex than the acoustic isolation chambers of traditional infant testing. The current series of studies used the headturn preference procedure (with video familiarization) to examine 7.5-month-old infants abilities to use visual/auditory correlations to reliably attend to and segment a given speech stream in the face of a distracting voice. Results indicated that in contrast to seeing a static face, infants succeeded at segmentation when a dynamic visual display of the face of the talker matched the acoustic passage. That is, when two blended voices were of equal loudness, infants could use visual correspondences to reliably recognize words presented in the matching video. Furthermore, they did so even if the video display was changed to a synchronized oscilloscope pattern, rather than a face. They also succeeded when the video display was simply a synchronized flashing light. These results suggest that this ability is related to domain-general sensitivities for intermodal correspondences, rather than any face-specific effects, and suggest a mechanism whereby face-specific effects could develop.

3aSC4. Two reasons not to bring your children to cocktail parties. Susan Nittrouer, Melanie Wilhelmsen, Kathy Shapley, Kathi Bodily (Ctr. for Persons with Disabilities, Utah State Univ., 6840 Old Main Hill, Logan, UT 84322), and Tom Creutz (Boys Town Natl. Res. Hospital, Omaha, NE 68131)

Two kinds of noise can affect speech perception at cocktail parties: others' speech (sloping spectrum) and the environment (level spectrum). Recognition scores for phonetically balanced monosyllabic words in level noise at five SNRs for adults and children (4 to 8 years) were collected, and compared to those of Nittrouer and Boothroyd for speech-shaped noise [J. Acoust. Soc. Am. **87**, 2705–2715 (1990)]. All listeners showed similar effects of level noise, and children's results matched those for speech-shaped noise. However, adults showed a 22% advantage for speech-shaped over level noise, suggesting they use high-frequency speech elements not masked by the sloping long-term spectrum of speech. Next labeling of words differing in syllable-final voicing (using vocalic duration and offset transitions, neither high-frequency cues) was examined for children and adults in quiet and in level noise. All listeners performed similarly in quiet. Adults' results for quiet and noise matched, but children showed a decreased weighting of transitions in noise compared to quiet, suggesting enhanced masking of this information. Thus children's speech perception in noise is impaired both because they fail to use high-frequency cues available against the background of others' speech and they experience enhanced masking of formant transitions, their preferred source of information. [Work supported by NIDCD Grant No. DC-00633.]

3aSC5. Effects of response format on speech intelligibility in noise: Results obtained from open-set, closed-set, and delayed response tasks. Cynthia G. Clopper, Adam T. Tierney, and David B. Pisoni (Speech Res. Lab., Dept. of Psych., Indiana Univ., Bloomington, IN 47405, cclopper@indiana.edu)

Many word recognition studies over the last 40 years have used forced-choice closed-set tasks, based on the assumption that closed-set and open-set tasks differ only in the level of chance performance. However, Sommers, Kirk, and Pisoni (1997) found that lexical competition and talker variability produce robust effects on performance only in open-set tasks, suggesting fundamental differences in the task demands and processing strategies between open- and closed-set tasks. In the present study, listeners were asked to recognize spoken words degraded by a bit-flipping algorithm in three response formats: open-set, closed-set "before," and closed-set "after." In the closed-set "before" condition, the six response alternatives were presented 1 s prior to the onset of the auditory signal. In the closed-set "after" condition, the response alternatives were presented 1 s after the auditory signal. Results revealed significant effects of lexical competition and talker variability only in the open-set task. These findings suggest that even a delay of 1 s is not adequate to induce the task demands observed in open-set word recognition tests. [Work supported by NIH.].

3aSC6. A test of the orthographic recoding hypothesis. Daniel E. Gaygen (Psych. Dept., Ithaca College, Ithaca, NY 14850, dgaygen@ithaca.edu)

The Orthographic Recoding Hypothesis [D. E. Gaygen and P. A. Luce, Percept. Psychophys. **60**, 465–483 (1998)] was tested. According to this hypothesis, listeners recognize spoken words heard for the first time by mapping them onto stored representations of the orthographic forms of the words. Listeners have a stable orthographic representation of words, but no phonological representation, when those words have been read frequently but never heard or spoken. Such may be the case for low frequency words such as jargon. Three experiments using visually and auditorily presented nonword stimuli tested this hypothesis. The first two experiments were explicit tests of memory (old–new tests) for words presented visually. In the first experiment, the recognition of auditorily presented nonwords was facilitated when they previously appeared on a visually presented list. The second experiment was similar, but included a concurrent articulation task during a visual word list presentation, thus preventing covert rehearsal of the nonwords. The results were similar to the first experiment. The third experiment was an indirect test of memory (auditory lexical decision task) for visually presented nonwords. Auditorily presented nonwords were identified as nonwords significantly more slowly if they had previously appeared on the visually presented list accompanied by a concurrent articulation task.

3aSC7. Perception of coda voicing from properties of the onset and nucleus of *led* and *let*. Sarah Hawkins (Dept. of Linguist., Univ. of Cambridge, Sidgwick Ave., Cambridge CB3 9DA, UK) and Noël Nguyen (Université de Provence, France)

Syllable-onset [l] is longer and often has different (usually lower) $F2$ frequency before a voiced coda. Five experiments (E1–E5) explore the perceptual power of these properties and $f0$. Listeners identified as *led* or *let* synthetic syllables whose latter part was replaced by noise 80 ms after vowel onset. The duration and $F2$ frequency of [l] steady-state were varied in E1–E5, $f0$ of [l] in E1, vowel formant frequencies in E2–E5, and stimulus randomization principles in E3–E5. In E1, [l] had one of six durations, two $F2$ frequencies, and two $f0$ starting frequencies. E2 continued $F2$ differences into the vowel, as in natural speech. E3–E5 varied $F2$ independently in [l] and vowel rather than together; stimuli were presented in one session, or blocked by vowel $F2$. Shorter [l]s, higher $f0$, and higher $F2$ in [l]+vowel produced more *let* responses. $F2$ in [l] (alone) mainly affected responses when vowel quality was constant. However, listeners learned which cues were systematic, and some who initially used $F2$ frequency switched to duration of [l] relatively late in a session. The results suggest coda voicing is a property of the whole syllable, and support word recognition models that are sensitive to systematic variation in subtle phonetic detail.

3aSC8. The effect of syllabification and gemination on $F2$ onsets in Swedish. Augustine Agwuele (Dept. of Linguist., Univ. of Texas, 1 University Station B5100, Austin, TX 78712)

The quest to explain the continuousness of speech on the physical level has been dominated by the co-production theory [Hman, J. Acoust. Soc. Am. **41** (1966)]. According to this view, the production of a VCV sequence involves a diphthongal movement from V1-to-V2, with a superimposed consonantal gesture. However, data from recent studies are at variance with this position—Modaressi [Ph.D. dissertation, UT Austin, 2002], Perkell [Coarticulation Strategies Speech Commun. **5**, 47–68 (1986)]. These studies document a trough-phenomena, which suggests a discontinuity in muscular activity during the production of the consonant. This paper provides additional acoustic evidence in support of sequential programming of consonant-vowel events as advocated by Joos [Acoustic Phonetics Lang. **24** (1948)]. It examines symmetrical VCV sequences in Swedish natural speech with syllable boundaries and duration of consonant gemination altered to produce different types of temporal interval between V1 and V2; i.e., V#CV, VCC#V, VC#CV, VCC#CV. $F2$ of V1mid, V1offset, V2onset, and V2mid were measured. Locus equations were plotted for all VC contacts. Statistical analysis of these data show: (1) de-activation of tongue movement at the CV boundary, (2) a reduction of the influence of V1 on V2 as a function of increasing consonant duration, and (3) a weak degree of CV coarticulation.

3aSC9. Perceptual discontinuities and categorization: Implications for speech perception. Lori L. Holt (Dept. of Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, lholt@andrew.cmu.edu), Andrew J. Lotto (Washington State Univ., Pullman, WA 99164), and Randy L. Diehl (Univ. of Texas, Austin, TX 78712)

Behavioral experiments with infants, adults and nonhuman animals converge with neurophysiological findings to suggest that there is a discontinuity in auditory processing of stimulus components differing in onset time by about 20 ms. This discontinuity has been implicated as a basis for boundaries between speech categories distinguished by VOT. Here, we investigate how this discontinuity interacts with the learning of novel perceptual categories. Adult listeners were trained to categorize a nonspeech acoustic cue that mimics the temporal distinction of VOT. One group of listeners learned categories with a boundary coincident with the perceptual discontinuity. Another group learned categories defined such that the perceptual discontinuity fell within a category. Listeners in the latter group required significantly more experience to reach criterion categorization performance. The evidence of interactions between the perceptual discontinuity and the learned categories extended to generalization tests as well. It has been hypothesized that languages make use of perceptual discontinuities to promote perceptual distinctiveness among sounds within a language inventory. The present data suggest that these influences interact with category learning. As such, learnability may play a predictive role in selection of language sound inventories. Moreover, it may be possible to observe predictable learning effects in infant speech perception.

3aSC10. Perceptual overshoot with speech and nonspeech sounds. Radhika Aravamudhan and John. W. Hawks (School of Speech Pathol. & Audiol., Kent State Univ., P.O. Box 5190, Kent State Univ., Kent, OH 44240, raravam1@kent.edu)

One of the basic quests in speech perception research has been to find the differences or similarities in the mechanisms involved in the perception of speech and nonspeech sounds. The current study will address the differences in perception of speech and nonspeech signals by comparing the perceptual overshoot in synthetic vowels and sinewave acoustic replicas of the synthetic vowels. Lindblom and Studdert-Kennedy (1967) demonstrated that the perceptual boundary for steady state vowels and that for vowels in a CV context with $F2$ transition are different. They called this phenomenon a perceptual compensation or perceptual overshoot. In the current study the perceptual boundaries for synthetic steady state vowels, steady state sinewave acoustic replicas of vowels, a vowel in the CV context with $F2$ transition and sinewave acoustic replicas of vowels in the CV context are compared. The results will be discussed in the poster. For Speech Communication Best Student Paper Award.

3aSC11. The effects of regional dialect on vowel intelligibility from a cross-linguistic perspective. Elaina M. Frieda (Dept of Psych., Auburn Univ., 226 Thach Hall, Auburn, AL 36894-5214, friedem@auburn.edu) and Robert A. Fox (The Ohio State Univ., Columbus, OH 43210)

The present experiment is a continuation of a previously reported study that examined intelligibility of English vowels as a function of dialect spoken by native speakers of English and Japanese. The purpose of this research is to assess how regional dialectal variations affect vowel intelligibility for native and non-native speakers of English. Native English and Japanese subjects were recorded in two divergent dialectal regions of the United States (Ohio and Alabama). These tokens were then employed in a perceptual experiment where native English and Japanese listeners from Ohio and Alabama identified the English vowels. To date, perceptual data from only Ohio native English and Japanese subjects have been reported. A further analysis of the data including Alabama native English and Japanese listeners revealed that native English speakers from Ohio obtained the highest intelligibility scores overall (for example, all four listener groups identified Ohio English more accurately than all other groups). Additionally, native Japanese speakers from Alabama received the lowest overall intelligibility scores. The tentative results of this study imply that non-native speakers of English that are exposed to a nonstandard dialect may have deleterious effects on comprehension.

3aSC12. The recognition of accented and unaccented English words by native speakers of Spanish and English. Satomi Imai, James Flege (Div. of Speech and Hearing Sci., Univ. of Alabama at Birmingham, CH20, 1530 3rd Ave. S., Birmingham, AL 35294, imais@uab.edu), and Amanda Walley (Univ. of Alabama at Birmingham, Birmingham, AL 35294)

This study examined effects of foreign accent and lexical factors (word frequency and neighborhood density) on the recognition of English words in noise. Two groups of native Spanish (NS) adults differing in overall degree of foreign accent (FA) in English (weaker versus stronger FAs) participated as well as a native English (NE) group. Participants identified words that had been spoken by a NE and a NS speaker. It was hypothesized that: (1) the stronger FA group would have less nativelike phonological representations than the weaker FA group, and so would benefit more from hearing Spanish-accented English words; (2) words in dense lexical neighbors would require finer discrimination of English sounds, necessitating more nativelike phonological representations. The results showed that the weaker FA group recognized as many low-neighborhood-density English words as the NE group. For high-neighborhood-density words, the weaker FA group recognized fewer unaccented English words than the NE group, but more accented English words. The stronger FA group recognized as many accented words as the other groups, but fewer

unaccented words. The differences between the weaker and stronger FA groups were interpreted as reflecting a change in their phonological representations for English words. [Work supported by NIH.]

3aSC13. The influence of linguistic experience on pitch perception in speech and nonspeech sounds. Tessa Bent, Ann R. Bradlow (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, t-bent@northwestern.edu), and Beverly A. Wright (Northwestern Univ., Evanston, IL 60208)

How does native language experience with a tone or nontone language influence pitch perception? To address this question 12 English and 13 Mandarin listeners participated in an experiment involving three tasks: (1) Mandarin tone identification—a clearly linguistic task where a strong effect of language background was expected, (2) pure-tone and pulse-train frequency discrimination—a clearly nonlinguistic auditory discrimination task where no effect of language background was expected, and (3) pitch glide identification—a nonlinguistic auditory categorization task where some effect of language background was expected. As anticipated, Mandarin listeners identified Mandarin tones significantly more accurately than English listeners (Task 1) and the two groups' pure-tone and pulse-train frequency discrimination thresholds did not differ (Task 2). For pitch glide identification (Task 3), Mandarin listeners made more identification errors: in comparison with English listeners, Mandarin listeners more frequently misidentified falling pitch glides as level, and more often misidentified level pitch ''glides'' with relatively high frequencies as rising and those with relatively low frequencies as falling. Thus, it appears that the effect of long-term linguistic experience can extend beyond lexical tone category identification in syllables to pitch class identification in certain nonspeech sounds. [Work supported by Sigma Xi and NIH.]

3aSC14. Perceptual prothesis in native Spanish speakers. Rachel M. Theodore and Anna M. Schmidt (School of Speech Pathol. and Audiol., Kent State Univ., Kent, OH 44240, rachel_theodore@yahoo.com)

Previous research suggests a perceptual bias exists for native phonotactics [D. Massaro and M. Cohen, Percept. Psychophys. **34**, 338–348 (1983)] such that listeners report nonexistent segments when listening to stimuli that violate native phonotactics [E. Dupoux, K. Kakehi, Y. Hirose, C. Pallier, and J. Mehler, J. Exp. Psychol.: Human Percept. Perform. **25**, 1568–1578 (1999)]. This study investigated how native-language experience affects second language processing, focusing on how native Spanish speakers perceive the English clusters /st/, /sp/, and /sk/, which represent phonotactically illegal forms in Spanish. To preserve native phonotactics, Spanish speakers often produce prothetic vowels before English words beginning with /s/ clusters. Is the influence of native phonotactics also present in the perception of illegal clusters? A stimuli continuum ranging from no vowel (e.g., ''sku'') to a full vowel (e.g., ''esku'') before the cluster was used. Four final vowel contexts were used for each cluster, resulting in 12 sCV and 12 VsCV nonword endpoints. English and Spanish listeners were asked to discriminate between pairs differing in vowel duration and to identify the presence or absence of a vowel before the cluster. Results will be discussed in terms of implications for theories of second language speech perception.

3aSC15. The representation of consonant clusters in the mental lexicon. Lisa J. Incognito and James R. Sawusch (Dept. of Psych., SUNY at Buffalo, Buffalo, NY 14260, incognit@acsu.buffalo.edu)

Previous work has shown that the perception of a phoneme in a syllable is influenced by the number of similar sounding words [lexical neighborhood (Newman, Sawusch, and Luce, 1997)]. This previous work determined neighborhoods for target syllables using a one phoneme change rule. For example, bow, bath and mouth are neighbors of bowth. The present work focused on how consonant clusters are represented in the mental lexicon. Nonsense syllables composed of initial consonant clusters followed by a vowel and final consonant were used as stimuli. Two rules were used to compute the neighborhood for each target syllable. One was the one phoneme change rule used in previous studies. The second treated clusters of consonants as single units in a one unit change rule. Target syllables with differential neighborhoods based on the two rules were the endpoints of the test series. Results to date agree with the one phoneme change rule. These results are consistent with models of word recognition which treat consonant clusters as a sequence of phonemes. [Work supported by NIDCD Grant R01DC00219 to SUNY at Buffalo.]

3aSC16. Sonority contours in word recognition. Sean McLennan (Indiana Univ., Memorial Hall, Rm. 322, 1021 E. Third St., Bloomington, IN 47405-7005, mmclenna@indiana.edu)

Contrary to the Generativist distinction between competence and performance which asserts that speech or perception errors are due to random, nonlinguistic factors, it seems likely that errors are principled and possibly governed by some of the same constraints as language. A preliminary investigation of errors modeled after the child's ''Chain Whisper'' game (a degraded stimulus task) suggests that a significant number of recognition errors can be characterized as an improvement in syllable sonority contour towards the linguistically least-marked, voiceless-stop-plus-vowel syllable. An independent study of sonority contours showed that approximately half of the English lexicon can be uniquely identified by their contour alone. Additionally, ''sororities'' (groups of words that share a single sonority contour), surprisingly, show no correlation to familiarity or frequency in either size or membership. Together these results imply that sonority contours may be an important factor in word recognition and in defining word ''neighborhoods.'' Moreover, they suggest that linguistic markedness constraints may be more prevalent in performance-related phenomena than previously accepted.

3aSC17. Neighborhood effects on speech-perception abilities of adults and children aged 6–9 years. Joan E. Sussman, Devon Dee, and Diana Curcio (Dept. of Communicative Disord. and Sci., Univ. at Buffalo, Buffalo, NY 14214)

According to Luce and Pisoni (1998), the mental lexicon is organized into sparse and dense neighborhoods. In perception studies with adults, typically words from sparse neighborhoods are more quickly identified than words from dense neighborhoods, likely due to a competition effect. In production however, words from dense neighborhoods are produced more quickly than words from sparse neighborhoods, probably due to a faciliation effect. In young children, these so-called ''neighborhood effects'' have been shown to be absent (Charles-Luce and Luce, 1990), related to the overall smaller size of the lexicon. The current investigation studied single-feature perception by 10 children, aged 6–9 years and 10 adults, looking at differences due to neighborhood density. Each target word was presented in one of two semantically neutral carrier sentences in three levels of babbled noise: no noise, +5-dB S/N ratio, and 0-dB S/N ratio (equal signal and noise levels). Results showed that children performed best in quiet and that words from dense neighborhoods were perceived more accurately than those from sparse neighborhoods except for the 0-dB condition. Adults surprisingly performed best in the +5-dB condition and a neighborhood effect shown only for the 0-dB condition.

3aSC18. Children's perception of static noise and static formant cues to stop-consonant place of articulation. Ralph N. Ohde (Dept. of Hearing and Speech Sci., Vanderbilt Bill Wilkerson Ctr., Vanderbilt Univ., Nashville, TN 37212-2197, ralph.n.ohde@vanderbilt.edu)

Children's processing strategies appear to favor dynamic cues such as formant transitions as compared to static cues such as $F2$ onsets and noise bursts. The purpose of this research was to examine children's perception of place of articulation based only on static cues. Ten children at each of five age levels (3, 4, 5, 6, and 7) and a control group of 10 adults identified

synthesized stop consonants [d g] in two vowel contexts [i ɑ]. The synthesis parameters included variations in $F2$ onsets and stop-consonant noise bursts. The $F2$ onsets were either ''appropriate'' or ''neutral'' for place of articulation. The noise bursts were either short (10 ms) or long (25 ms). Preliminary data show that the $F2$ onset is not as salient in children's perception as in adults' perception. In addition, children more often than adults categorized neutral $F2$ onset stimuli as ambiguous indicating stronger category formation in the latter than former groups. The role of noise bursts was more salient in adult perception than child perception. The findings will provide information on the role of ''static'' cues, on the perceptual integration of ''static'' noise and formant cues, and on the influence of sound category formation in perceptual development. [Work supported by NIH and a Vanderbilt University Research Council Grant.]

3aSC19. Cue weighting of static and dynamic vowel properties in children versus adults. Sarah R. Malech and Ralph N. Ohde (Dept. of Hearing and Speech Sci., Vanderbilt Bill Wilkerson Ctr., Vanderbilt Univ., Nashville, TN 37212-2197, ralph.n.ohde@vanderbilt.edu)

The purpose of this study was to determine whether children give more perceptual weight than do adults to dynamic spectral cues versus static cues, when identifying vowel sounds. Three experimental stimulus sets were presented, each with 30-ms stimuli. The first consisted of unchanging formant onset frequencies ranging in value from frequencies for [i] to those for [ɑ], corresponding to a bilabial stop consonant. The second two consisted of either an [i] or [ɑ] onset frequency with a 25-ms portion of a formant transition whose trajectory was toward one of a series of target frequencies ranging from those for [i] to those for [ɑ]. Ten children between the ages of 3;8 and 4;1 and a control group of 10 adults identified each stimulus as [bi] or [bɑ]. The results showed developmental effects: the children relied more heavily than the adults did on the static formant onset frequency cue to identify the vowels, while the adults appeared to give more equal weight to both static and dynamic cues than the children did. These findings contradict the Developmental Perceptual Weighting Shift theory and are discussed in relation to this theory and other current research on the development of vowel perception.

3aSC20. Perception of place-of-articulation information by monkeys versus humans. Joan M. Sinnott and Casey S. Gilmore (Comparative Hearing Lab, Psych. Dept., Univ. of South Alabama, Mobile, AL 36688)

Four monkeys and six humans representing five different native languages were compared in the ability to categorize natural CV tokens of /b/ vs /d/ produced by four talkers of American-English (two male; two female) in four vowel contexts /i,e,a,u/. A two-choice left/right procedure was used in which percent correct and response time data were compared between species. Both measures indicated striking vowel context effects for monkeys, but none for humans. Specifically, monkeys performed better for back vowels /a,u/ than front vowels /i,e/. Since back vowels have more distinctive $F2$ onset transitions differentiating /b/ vs /d/, these results imply that monkey perception is more dependent than human perception on the actual acoustic structure of the syllables. We conclude that humans do not use general mechanisms in place perception, rather they use some sort of special mechanism to eliminate vowel context effects. While monkeys do not provide accurate models of adult humans, they may be able to provide a model of the preverbal human infant before it learns a more speech-specific adult strategy of place information extraction. [Work supported by NIH.]

3aSC21. The role of air pressure and contact force in shaping obstruent consonant onset. Lan Chen (Speech Commun. Group, Res. Lab of Electronics, MIT, 77 Massachusetts Ave., Boston, MA)

Soft tissues (the tongue or lips) are used to form the narrow oral constriction for turbulence noise generation during the production of obstruent consonants. The displacement of the soft tissue subject to oral pressure buildup is comparable to the vertical dimension of the constriction. The contact force during the closure of stop consonants and affricates provides a pressure load over 5 times larger than the air pressure at the surface in contact. It can influence the time variation of the constriction size at onset in the form of elastic energy stored in the compliant structure forming the constriction. A finite element fluid–structure interaction program has been used to simulate the effect of these external forces during the onset of obstruent consonants. Preliminary results from a 2-D tongue tip constriction/closure model show that air pressure and contact force can introduce movement on the order of 0.1–0.2 mm during the first tens of milliseconds after release, which is enough to affect the size of the constriction at onset and the nature of release burst. The results of this kind can be used for speech synthesis, guiding the modification of the trajectories of articulators at the consonant onset. [Work supported by NIH.]

3aSC22. A labial gesture for /l/. Fiona Campbell (Dept. of Linguist., Univ. of British Columbia, E270-1866 Main Mall, Vancouver, BC V6T 1Z1, Canada, fionaca@interchange.ubc.ca) and Bryan Gick (Univ. of British Columbia, Vancouver, Canada)

Both in language change and in substitutions during language acquisition and disordered speech, /l/ has often been observed to alternate with labial sounds such as [w] or rounded vowels, particularly in postvocalic position. While there are many possible explanations for this alternation, including acoustic enhancement and articulator coupling, one possibility that has not been tested is whether normal adult speakers of English actually produce lip rounding for /l/. A study was conducted to test for the presence of a labial gesture in normal productions of /l/. Front and side video data of lip positions were collected from three adult English speakers during productions of /l/ and /d/. Significant differences were found for all subjects in lip protrusion (upper and lower) and/or lip aperture (horizontal and vertical) in post-vocalic allophones, as well as between the pre- and post-vocalic allophones of /l/. No significant differences were observed in comparisons of pre-vocalic /l/ and /d/. Results suggest that there is in fact a labial gesture in the post-vocalic allophone of /l/, but not in the pre-vocalic allophone. These findings are consistent with a notion of gestural simplification as a possible explanation for substitutions and in language change. [Research supported by NSERC.]

3aSC23. Language-specific vocal tract configurations during nonspeech. Bryan Gick (Dept. of Ling., Univ. of British Columbia, E270-1866 Main Mall, Vancouver, BC V6T 1Z1, Canada) and Clare Cook (Univ. of British Columbia, Vancouver, Canada)

Previous work has been found to be surprisingly low within-speaker variability in baseline articulator positions during inter-utterance nonspeech [Gick, Phonetica (2002)], raising the question of whether these baseline positions may in fact be active in speech production. If so, then they should be specified and should vary systematically across languages. A study was conducted to test for cross-language differences in inter-utterance articulator positions. Individual video frames were extracted at the midpoint of interutterance pauses in x-ray films of 5 French and 5 English speakers. Measures were made of articulator positions relative to fixed bone points, and values normalized to jaw size. Frames with potentially confounding surrounding phonetic contexts were omitted. Results for lip measures indicate that French speakers have significantly greater protrusion of the lower lip, but significantly less upper lip protrusion, than English speakers. Additional results will be presented for lingual articulators. Thus these baseline vocal tract configurations do appear to be specified differently for different languages. Additional implications will be discussed, such as possible roles these configurations may play in phonology, potential influence on vowel systems (especially schwa), and cross-language vowel normalization. [Research supported by NSERC and NIH.]

3aSC24. Lip interactions and closure duration in labial consonants. Anders Lofqvist (Haskins Labs., 270 Crown St., New Haven, CT 06511)

This study examines interactions between the upper and lower lips in labial consonants where the duration of the oral closure is varied for linguistic purposes. Earlier work has shown that such interactions occur and that their magnitude is partly related to the duration of the oral closure. Lip movements were recorded in native Japanese speakers using a magnetometer system. Results show a positive correlation between the vertical positions of the upper and lower, when both are measured at the point in time where the lower lip reached its peak position during the oral closure. Since the peak vertical position of the lower lip increased with closure duration, the upper lip also had a higher vertical position at the same point in time for long than for short consonants. During the oral closure, the lower lip continued to move upward with a larger movement for long than for short consonants. Due to the mechanical interaction between the lips, the upper lip reversed its downward movement at the onset of the oral closure and also moved upward during the closure, again with a larger movement for long than for short consonants. [Work supported by NIH.]

3aSC25. When there is way too much stress: A first look at the failure behavior of vocal fold tissue. Roger W. Chan (Vocal Fold Physiol. and Biomechanics Lab., Dept. of Audiol. and Speech Sci., Purdue Univ., West Lafayette, IN 47907) and Thomas Siegmund (Purdue Univ., West Lafayette, IN 47907)

In normal voice production, linear small-amplitude vocal fold oscillation occurs only under restricted conditions, such as during oscillation onset and offset. More often than not, phonation in physiological range involves large-amplitude oscillation that is associated with the development of tissue shear stresses and strains much beyond their linear viscoelastic limit, particularly in the vibrating portion of the vocal fold, i.e., the extracellular matrix (ECM). This study targeted one such large-strain nonlinear viscoelastic behavior of the vocal fold by attempting to quantify the stress failure response of the vocal fold ECM under shear. Sheep vocal fold ECM specimens were subjected to torsional, steady shear in a controlled-strain rheometer *in vitro*, at constant strain rates of 0.01, 0.1, and 1.0 rad/s. Results showed that the vocal fold ECM demonstrated nonlinear stress–strain response, as well as failure response when shear strain reached around 100%. Strain-dependent and rate-dependent onset of partial and complete stress failure was observed. A constitutive approach based on a standard-linear cohesive zone model was formulated to characterize the observed rate-dependent failure behavior of the vocal fold ECM. These findings have important implications for predicting tissue injury and for establishing stress and strain safety limits for large-amplitude vocal fold oscillation.

3aSC26. The effect of physical constraints on articulatory variability in English /r/. Jason Brown (Dept. of Linguist., Univ. of British Columbia, E270-1866 Main Mall, Vancouver, BC V6T 1Z1, Canada) and Bryan Gick (Univ. of British Columbia, Vancouver, BC V6T 1Z1, Canada and Haskins Labs., New Haven, CT)

In a recent study of American English /r/, Guenther *et al.* [J. Acoust. Soc. Am. (1999)] hypothesized that articulatory ''tradeoff'' correlations are the result of the need to maintain stable acoustic targets. Their findings included (1) a positive correlation between tongue back height and tongue front horizontal position for 7/7 subjects, (2) a negative correlation between tongue back height and tongue front height for 5/7 subjects, and (3) a positive correlation between tongue front horizontal position and tongue front height for 2/7 subjects. The present study investigates the possibility that these correlations result from physical constraints on the tongue such as volume preservation and palate angle. Continuous sentences from the Wisconsin x-ray microbeam database were analyzed to determine whether these same correlations were present across whole utterances presumably lacking a stable $F3$ target. Results to date show some significant correlations despite extremely high noise levels. Results will be presented for additional measures using vowels only to reduce noise. These initial results suggest that the observed correlations may not result from strict acoustic targets for /r/, but rather from internal and external physical constraints on the tongue. [Work supported by NSERC and NIH.]

3aSC27. An articulatory examination of variable word-final flapping at phrase edges and interiors. Teruhiko Fukaya and Dani Byrd (USC Linguist., 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693)

Formulations of flapping as a symbolic phonological rule suggest clear articulatory differences between flaps and stops and offer no overt explanation for why phrase boundaries should block the alternation. A current articulatory phonetic study of word-final coronals by de Jong (1998) explored the possibility that gradient changes in articulatory dimensions might give rise to quantal difference in the percept of these coronal stops. In particular, increased overlap between the coronal and the following vowel may result in spatial reduction of the coronal gesture via blending. We present results from an experiment with three speakers examining tongue tip movement and acoustic data for word-final [t], phrase internally and at phrase boundaries, with falling and level stress contours. The results confirm that flaps did not occur at phrase boundaries. Phrase-internally, flaps occurred for one subject invariantly and for another subject variably. While the acoustic durations of flaps were expectedly shorter, this did not correspond directly to their closing movement durations. There was also some tendency for flaps to have a shorter acceleration phase and no change in deceleration duration, arguing against an overlap account. Finally, the phrase boundary was associated with articulatory and acoustic lengthening, both incompatible with patterns seen for flapped tokens. [Work supported by USC Zumburge Grant and by NIH.]

3aSC28. An approach to real-time magnetic resonance imaging for speech production. Shrikanth Narayanan (Dept. of Elec. Eng./Linguist., Univ. of Southern California, Los Angeles, CA 90089, shri@sipi.usc.edu), Krishna Nayak (Stanford Univ.), Dani Byrd, and Sungbok Lee (Univ. of Southern California)

Magnetic resonance imaging has served as a valuable tool for studying primarily static postures in speech production. Now, recent improvements in imaging techniques, particularly in temporal resolution, are making it possible to examine the dynamics of vocal tract shaping during speech. Examples include Mady *et al.* (2001, 2002) (8 images/second, T1 fast gradient echo) and Demolin *et al.* (2000) (4–5 images/second, ultra fast turbo spin echo sequence). The present study uses a non 2D-FFT acquisition strategy (spiral *k*-space trajectory) on a GE Signa 1.5T CV/i scanner with a low-flip angle spiral gradient echo originally developed for cardiac imaging [Kerr *et al.* (1997), Nayak *et al.* (2001)] with reconstruction rates of 8–10 images/second. The experimental stimuli included English sentences varying the syllable position of /n, r, l/ (spoken by 2 subjects) and Tamil sentences varying among five liquids (spoken by one subject). The imaging parameters were the following: 15 deg flip angle, 20-interleaves, 6.7 ms TR, 1.88 mm resolution over a 20 cm FOV, 5 mm slice thickness, and 2.4 ms spiral readouts. Data show clear real-time movements of the lips, tongue and velum. Sample movies and data analysis strategies will be presented. Segmental durations, positions, and inter-articulator timing can all be quantitatively evaluated. [Work supported by NIH.]

3aSC29. Speech task and timing considerations in MRI research. Melissa A. Epstein and Maureen Stone (Univ. of Maryland Dental School, 666 W. Baltimore St., Baltimore, MD 21201)

In order to create dynamic magnetic resonance images, subjects must repeat tokens as much as 30 times in a row, therefore, their precision is critical for image quality. Word repetition rate can be 1 or 1.5 s, and is matched to the rate of image recording. Unlike reciting tokens to the beat of a metronome, subjects must say each token during an extended period of noise that lasts about 75% of the repetition period. Furthermore, some tokens may be more difficult to say repetitively in these conditions. Therefore, this study simulated the acoustics of a MRI recording session to examine the effects of subject, token, and repetition rate on temporal precision. Subjects repeat up to ten mono- to trisyllabic words 30 times each at two repetition rates. Measurements are made of onsets and offsets of one or more phonemes within each word. Preliminary results (five subjects) indicate that there is an effect of subject and word. Further subjects are being measured to corroborate these results and to determine if one repetition rate is better than another and to test subject reliability.

3aSC30. Hearing smiles and smile suppression in natural speech. Amy K. Drahota and Vasudevi Reddy (Dept. of Psych., Univ. of Portsmouth, King Henry Bldg., King Henry I St., Portsmouth, Hants PO1 2DY, UK)

That we can hear smiles in speech is an established finding. However smiles in natural speech can be of many different kinds, serving different social functions. Previous research has focused only on one category of smile using either smiles ''posed'' during speech or degraded samples of smiling speech (to disguise the content of utterances). The present study used naturally occurring speech in three foreign languages (Czech, Spanish, and Finnish) and in English, presented to naive native English speakers. Preliminary analyses extracted two kinds of smiles in speech—''open smiles'' and ''suppressed smiles'' in contrast to speech with ''no smiles.'' Eighty listeners were presented with 18 audio-clips (six of each type) in randomized order. ''No smiles'' and ''open smiles'' were successfully identified across all languages. ''Suppressed smiles'' were most often coded as ''no smiles.'' An exploration of the pitch contours showed that ''suppressed smiles'' had a higher average pitch similar to ''open smiles,'' but a significantly higher variance in pitch range than others. Natural speech produces smiles serving very different functions. The auditory expression and perceivability of emotion is likely to be influenced by subtle social functions only evident in natural interactions.

3aSC31. Effects of the age of cochlear implantation on the quality of the speech produced by profoundly HOH speakers. Samantha Lake and Betty Kollia (Dept. of Commun. Disord., William Paterson Univ., Wayne, NJ 07470)

Four hearing-impaired children with prelingual, bilateral, severe-to-profound hearing loss were grouped by age, gender, and age at implantation; the younger group consisted of females approximately 6 years old and implanted between 1–2 years of age and the older group consisted of males approximately 14 years old and implanted at 9 years of age. Each child was diagnosed with prelingual hearing loss, was implanted with the Nucleus 24® cochlear implant in 1998, and has approximately 4 years of experience using the implant consistently. All subjects receive 8 h of direct instruction with the implant per week, in a school for the deaf that utilizes total communication. Each subject also receives speech therapy in 30-min sessions four times per week and exhibits intelligible speech. Coarticulation in the children's speech was studied using five consonant-diphthong-consonant pseudowords, in the carrier sentence ''it's a —— again.'' The recordings were digitized and analyzed acoustically. The results are discussed with reference to the age of cochlear implantation of the children and its role in the quality of their speech.

3aSC32. The effect of spectrally and temporally altered auditory feedback on speech intonation by hard of hearing listeners. Dragana Barac-Cikoja, Chizuko Tamaki, and Lannie Thomas (Gallaudet Univ., MTB B09, 800 Florida Ave. NE, Washington, DC 20002)

Eight listeners with severe to profound hearing loss read a six-sentence passage under spectrally altered and/or delayed auditory feedback. Spectral manipulation was implemented by filtering the speech signal into either one or four frequency bands, extracting respective amplitude envelope(s), and amplitude-modulating the corresponding noise band(s). Thus, the resulting auditory feedback did not preserve intonation information, although the four-band noise signal remained intelligible. The two noise conditions and the unaltered speech were each tested under the simultaneous and three delayed (50 ms, 100 ms, 200 ms) feedback conditions. Auditory feedback was presented via insert earphones at the listener's most comfortable level. Recorded speech was analyzed for the form and domain of the fundamental frequency ($f0$) declination, the magnitude of the sentence initial $f0$ peak (P1), and the fall–rise pattern of $f0$ at the phrasal boundaries. A significant interaction between the two feedback manipulations was found. Intonation characteristics were affected by speech delay only under the spectrally unaltered feedback: The magnitude of P1 and the slope of the $f0$ topline both increased with the delay. The spectral smearing diminished the fall–rise pattern within a sentence. Individual differences in the magnitude of these effects were significant.

3aSC33. The impact of rate reduction and increased vocal intensity on coarticulation in dysarthria. Kris Tjaden (Dept. of Communicative Disord. & Sci., Univ. at Buffalo, 122 Cary Hall, 3435 Main St., Buffalo, NY 14214-3005, tjaden@acsu.buffalo.edu)

The dysarthrias are a group of speech disorders resulting from impairment to nervous system structures important for the motor execution of speech. Although numerous studies have examined how dysarthria impacts articulatory movements or changes in vocal tract shape, few studies of dysarthria consider that articulatory events and their acoustic consequences overlap or are coarticulated in connected speech. The impact of rate, loudness, and clarity on coarticulatory patterns in dysarthria also are poorly understood, although these prosodic manipulations frequently are employed as therapy strategies to improve intelligibility in dysarthria and also are known to affect coarticulatory patterns for at least some neurologically healthy speakers. The current study examined the effects of slowed rate and increased vocal intensity on anticipatory coarticulation for speakers with dysarthria secondary to Multiple Sclerosis (MS), as inferred from the acoustic signal. Healthy speakers were studied for comparison purposes. Three repetitions of twelve target words embedded in the carrier phrase ''It's a —— again'' were produced in habitual, loud, and slow speaking conditions. $F2$ frequencies and first moment coefficients were used to infer coarticulation. Both group and individual speaker trends will be examined in the data analyses.

3aSC34. Adductor spasmodic dysphonia: Relationships between acoustic indices and perceptual judgments. Michael P. Cannito (School of Audiol. & Speech Pathol., Univ. of Memphis, 807 Jefferson Ave., Memphis, TN 38105, mcannito@memphis.edu), Christine M. Sapienza, Gayle Woodson (Univ. of Florida, Gainesville, FL), and Thomas Murry (Columbia Univ., New York, NY)

This study investigated relationships between acoustical indices of spasmodic dysphonia and perceptual scaling judgments of voice attributes made by expert listeners. Audio-recordings of The Rainbow Passage were obtained from thirty one speakers with spasmodic dysphonia before and after a BOTOX injection of the vocal folds. Six temporal acoustic measures were obtained across 15 words excerpted from each reading sample, including both frequency of occurrence and percent time for (1) aperiodic phonation, (2) phonation breaks, and (3) fundamental frequency shifts. Visual analog scaling judgments were also obtained from six voice experts using an interactive computer interface to quantify four voice attributes (i.e., overall quality, roughness, brokenness, breathiness) in a carefully psychoacoustically controlled environment, using the same reading passages as stimuli. Number and percent aperiodicity and phonation breaks correlated significanly with perceived overall voice quality, roughness, and brokenness before and after the BOTOX injection. Breathiness was correlated with aperidocity only prior to injection, while roughness also correlated with frequency shifts following injection. Factor analysis reduced perceived attributes to two principal components: glottal squeezing and breathiness. The acoustic measures demonstrated a strong regression relationship with perceived glottal squeezing, but no regression relationship with breathiness was observed. Implications for an analysis of pathologic voices will be discussed.

3aSC35. Open-source software for speech perception research. Robert T. Gayvert (Gayvert Consulting, 16 Chase View Rd., Fairport, NY 14450) and James M. Hillenbrand (Western Michigan Univ., Kalamazoo, MI 49008)

The purpose of this paper is to describe some relatively simple software that can be used for performing such routine tasks as controlling listening experiments (e.g., simple labeling, discrimination using procedures such as ABX, oddity, same–different, etc., sentence intelligibility, magnitude estimation, and so on), recording responses and response latencies, analyzing and plotting the results of those experiments, displaying instructions, and making scripted audio recordings. The software runs under Windows and is controlled by creating text files that allow the experimenter to specify key features of the experiment such as the stimuli that are to be presented, the randomization scheme, inter-stimulus and intertrial intervals, the format of the output file, and the layout of response alternatives on the screen. Some simple demonstrations will be provided, along with instructions for downloading the software. [Work supported by NIH.]

WEDNESDAY MORNING, 30 APRIL 2003 ROOM 203, 10:00 A.M. TO 12:00 NOON

Session 3aSP

Signal Processing in Acoustics: General Topics on Signal Processing in Acoustics

Ning Xiang, Cochair

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Ronald A. Wagstaff, Cochair

National Center for Physical Acoustics, University of Mississippi, P.O. Box 1848, University, Mississippi 38677-1848

Contributed Papers

10:00

3aSP1. Time-reversal maximum-length sequence pairs for simultaneous acoustical dual source measurements. Ning Xiang, Richard Raspet, and Kevin Dillion (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr. University, MS 38677)

A binary sequence generated by a linear shift-register (maximum-length sequence, M-sequence) and its reversed-order sequence form a time-reversal (reciprocal) M-sequence pair. Their correlation property includes a two-valued pulse-like autocorrelation function and a relatively smaller-valued crosscorrelation function. This unique property, along with other number-theory properties, makes the time-reversal MLS pairs suitable for simultaneous dual source crosscorrelation measurements. In the simultaneous measurement of a dual source system, each of the time-reversal MLS pair simultaneously excites one of two separate sources, one or several receiver signals cross-correlate in turn with each of the MLS pairs resulting in impulse responses associated with two separate sources. The proposed method is particularly valuable for system identification tasks with multiple sound/vibration sources and receivers that have to be accomplished in a limited time period. A fast algorithm: fast MLS transform pair is developed for the crosscorrelation. Its feasibility and potential applications in the acoustical measurements are demonstrated using recent field experimental results.

10:15

3aSP2. Modeling and stability analysis of a dynamic system using control theory and signal processing approaches. Ali T. Herfat (Emerson-Copeland Corp., 1675 W. Campbell Rd., Sidney, OH 45365)

Presented in this paper are the following: Analog and digital modeling of a feedback control system; the system stability analysis, using analog and digital methods; and designing the digital system to meet steady-state error and transient response specifications. The digital control systems can control numerous loops at a reduced cost. System modifications can be implemented with software changes rather than hardware changes. Typically, the digital computer is placed in the forward control path, and modeled as a sample and hold network. Digital-to-analog and analog-to-digital converters are required within the system to ensure compatibility of the analog and digital signals. As we proceed with the analysis along with a case study, these criteria will be become apparent in this paper. An industrial dynamic system will be presented as a Dynamic Control System (the case study) in this paper.

10:30

3aSP3. Phase alignment for coherent and vector processing. Ronald A. Wagstaff (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

Acoustic phase angle is seldom used to achieve signal-to-noise ratio (SNR) gain for single sensor and beam spectral output. One reason is that phase angles generally progress at a nonuniform rate, with unpredictable changes in the direction of rotation. This causes fluctuations in the phase angles with corresponding reductions in gain, often including severe attenuation and cancellation of signals. By adopting a particular analytically convenient definition for phase fluctuations, the fluctuations, thus defined, constitute a set of aligned-phase angles. The aligned-phase angles can be used instead of phase angles to form phase-aligned coherent and vector averages. Doing so achieves SNR gains that equal or exceed the theoretical value of 10 log(N) for perfectly coherent vector averaging (N is the number of elements averaged). This is accomplished without the signal attenuation and cancellation common to coherent and vector averaging. Furthermore, the aligned-phase angles can also be used to automatically detect signals, based on both phase-aligned coherence and appropriate averaging of the aligned-phase angles. Results are included for wind noise in outdoor measurements. [Work supported by U.S. Army Space and Missile Defense Command.]

10:45

3aSP4. Application of discrete wavelet transform to measurement of traffic using road environmental sound from microphone on vehicle. Hironori Hara and Shinji Ozawa (Dept. of Information and Computer Sci., Keio Univ., Japan, 3-14-1 Hiyoshi, Kohoku-ku, Yokohama, Kanagawa 223-8522, Japan)

It is important for progress of ITS technology to develop the sensor that receives surrounding information and algorithm that extracts information. Recently the media generally used for ITS research is a picture, and not much research using sound is done. But sound sensor is still expected because sound has global information and less calculation cost than a picture. And research of a probe car that collects surrounding situations by the sensor is generally done. So improvement in sensing technology is desired. Moreover discrete wavelet transform attracts attention as the technique of signal processing. DWT is suitable for the analysis of nonstationary signals as road environmental sound. So the new system of the probe car using a sound sensor is proposed. Measurement of traffic is assumed as a probe car application example. In this paper, the result of count passing oncoming car with DWT to road environmental sound is reported. Road environmental sound was recorded in two kinds of road, uncovered road and tunnel, the validity of algorithm was verified by experiment.

11:00

3aSP5. Influence of spectral shape on ordinary and higher order correlation detection of broadband transients with prefiltering. Marcella E. Dean, George E. Ioup, Juliette W. Ioup (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148), and Lisa A. Pflug (Stennis Space Center, MS 39529)

The performances of ordinary and higher order correlation detectors of broadband chirps, amplitude modulated by a Gaussian envelope or a cosine taper and of various center frequencies and widths, are compared. It has been previously reported that using several inputs to the detectors, including functionals such as the autocorrelation, the energy spectrum, and wavelet transforms of the signals, provided no significant improvement over ordinary correlation detection. It was only after prefiltering the signal to some fraction of its passband (a partially-known source scenario, i.e., only the passband is known) that quite significant gains were achieved. These gains, up to 12.5 dB in the signal-to-noise ratio at the minimum detectable level, were obtained for Gaussian-modulated chirps. These chirps have spectra which are fairly sharply peaked in their passbands. For comparison, the detection performance for cosine-tapered chirps, whose spectrum is fairly flat in the passband, is examined. Performance gains are not as large, but still reach almost 10 dB. In all partially-known prefiltering, the tricorrelation detector has the best performance, which is achieved using the received signal input for the Gaussian-modulated chirp and the autocorrelation input for the cosine-tapered chirp. The known source performance is also investigated. [Research supported by ONR.]

11:15

3aSP6. Spatio-temporal gradient signal processing for detecting subsurface cracks. Kenbu Teramoto (Dept. of Mech. Eng., SAGA Univ., Saga-shi 8408502, Japan) and Kohsuke Tsuruta

This paper presents a crack characterization method by spatio-temporal gradient analysis over the Lamb wave field. The surface ultrasound waves that propagate in the direction of the layer are potential candidates of nondestructive testing (NDT) methodology. The proposed method has an ability to classify the surface acoustic wave field through the rank of the covariance matrix defined over the four-dimensional vector space which is spanned by the following components: a vertical displacement, its vertical velocity, and a pair of shear strains of the surface. The covariance matrix provides the information about cracks. The determinant of the covariance matrix, therefore, is proposed as the index of homogeneity of the object surface. In this study, the computational process in the wave field near the cracks is discussed and their physical meanings are investigated through FDTD-simulations and acoustic experiments.

11:30

3aSP7. Simultaneous detection and parameter estimation of multiple nonlinear acoustic chirps. Jasdeep S. Dhanoa, Evan J. Hughes, and Richard F. Ormondroyd (Cranfield Univ., Royal Military College of Sci., Shrivenham, Swindon SN6 8LA, UK)

Historically, short-time spectral analysis using the Fourier transform and its variants has been the primary method for time-frequency analysis. Recently, Wigner-based methods which provide time-frequency distributions have been applied to linear chirp detection. These methods are limited by the effects of windowing and, for the Wigner methods, cross-spectral components. Furthermore, they do not characterize the signal parameters directly but this may be accomplished by applying a Hough or Radon transform to the time-frequency distribution. For the detection of nonlinear chirps, these methods would be computationally intensive and have low accuracy. This paper introduces a new technique based on evolutionary algorithms for direct detection and parametrization of multiple nonlinear chirps within a window of observation. The optimization of the estimated chirp parameters is done in a nonlinear process using the evolutionary algorithm and the performance of the algorithm is not affected by peak broadening or the cross-spectral components. It exploits the windowing of the finite duration signal and provides higher accuracy even under conditions of high noise. The method also detects pure sinusoids and multiple linear chirps implicitly. The output parameters are, the start and stop frequencies, the phase, amplitude and the coefficients of the nonlinear variation of frequency with time.

11:45

3aSP8. Development of an underwater target classifier using target specific features. M. H. Supriya and P. R. Saseendran Pillai (Dept. of Electron., Cochin Univ. of Sci. and Technol. Kochi, India)

In Sonar, the detection and estimation functions are performed by signal processors, which involve the computation of various statistics, for enhancing the overall performance of the system. This also takes into account all the undesirable propagation effects caused by the underwater channel. Underwater targets can be classified by using certain target specific features such as target strength, target dynamics, and the signatures of the noise generated by the targets. Rough identification of the targets is carried out with target strength values at known aspects while for precise identification, classification clues from target dynamics and target signatures are generated. Databases for the engine noise spectra of various underwater targets, propeller noises, machinery noises and cavitation noises, speed-noise characteristics, etc., have been developed. The signal energy estimated within a finite-time interval is compared with the earlier detection/estimation decisions, which are stored in the target data record and the relevant target data are updated. The algorithm for identification of target from the most matching signature patterns in the database will generate the classification clues, which will help in target identification. Salient highlights of an underwater target classifier using the above-discussed target specific features are presented in this paper.

Session 3aUW

Underwater Acoustics, Signal Processing in Acoustics and Engineering Acoustics: Robust Passive Sonar II

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Brian H. Tracey, Cochair
Lincoln Laboratory, Massachusetts Institute of Technology, 244 Wood Street, Lexington, Massachusetts 02173-6426

Chair's Introduction—8:00

Invited Papers

8:05

3aUW1. From acoustic observatories to robust passive sonar. Arthur B. Baggeroer (MIT, Cambridge, MA 02139)

The evolution of the DARPA Robust Passive Sonar (RPS) as well as the ONR Shallow Water Acoustic Testbed (SWAT) programs can be traced from concept of an acoustic observatory posed by Munk in 1980 through several assessment and feasibility studies to their current implementations. During this, the thinking on several key hypotheses matured. (i) Are noise fields directional enough to sustain high array gains? (ii) What are the tradeoffs among nonstationarity caused by ship motion, array configuration (geometry and the number of sensors), and ''snapshots'' needed for stable adaptive processing? (iii) What is the interaction between gains from vertical and horizontal apertures? (iv) How much signal gain degradation is acceptable? (v) What methods of post-processing can be done for normalization, tracking, and 3-D localization? This presentation will give a brief summary of the history of RPS and SWAT and pose the question of how well we can answer some of hypotheses which motivated them.

8:30

3aUW2. Some approaches to robust, snapshot-deficient, adaptive processing. Heechun Song (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA 92093-0238)

We typically assume that longer arrays and more sensors are always beneficial. However, one of the issues identified by the ONR Ocean Acoustic Observatory Panel was ''snapshot-deficient adaptive processing'' arising, for example, from the motion of strong interferers (e.g., ships) through the small resolution cells of large arrays. This paper discusses two different but related concepts to reject the moving interferers: (1) null broadening for individual nulling and (2) model-based projection for collective nulling. The plane wave null broadening approach is combined with the robust white noise constraint and extended to matched field processing. The model-based projection exploits the fact that most interferers are close to the surface while targets of interest are usually in the middle of the water column. The projection matrix is constructed from the most significant eigenvectors of a replica covariance matrix from many independent sources within a surface strip. Simulations and experimental data illustrate the techniques.

8:55

3aUW3. Approaches to robustness. Henry Cox and Kevin D. Heaney (Orincon Corp., 4350 N. Fairfax Dr., Ste. 470, Arlington, VA 22203)

The term robustness in signal processing applications usually refers to approaches that are not degraded significantly when the assumptions that were invoked in defining the processing algorithm are no longer valid. Highly tuned algorithms that fall apart in real-world conditions are useless. The classic example is super-directive arrays of closely spaced elements. The very narrow beams and high directivity could be predicted under ideal conditions, could not be achieved under realistic conditions of amplitude, phase and position errors. The robust design tries to take into account the real environment as part of the optimization problem. This problem led to the introduction of the white noise gain constraint and diagonal loading in adaptive beam forming. Multiple linear constraints have been introduced in pursuit of robustness. Sonar systems such as towed arrays operate in less than ideal conditions, making robustness a concern. A special problem in sonar systems is failed array elements. This leads to severe degradation in beam patterns and bearing response patterns. Another robustness issue arises in matched field processing that uses an acoustic propagation model in the beamforming. Knowledge of the environmental parameters is usually limited. This paper reviews the various approaches to achieving robustness in sonar systems.

9:20

3aUW4. Motion compensation for adaptive horizontal line array processing: A beam domain approach. T. C. Yang (Naval Res. Lab., Washington, DC 20375)

Large aperture horizontal line arrays have small resolution cells and can be used to separate a target signal from an interference signal by array beamforming. High-resolution adaptive array processing can be used to place a null at the interference signal. But these features are significantly degraded by the source motion, which reduces the time period under which the environment can be considered stationary. For adaptive array processing, a large number of data samples are generally required to minimize the variance of the cross-spectral density between the array elements. The penalty of integrating over a large number of samples, when the source and/or interference moves, is the spread of signal and interference energy to many eigenvalues and consequently, the ability to suppress the interference suffers. We adapt a beam domain approach to compensate for the source motion allowing the beam covariance matrix of the signal to be integrated over a large number of data samples without signal energy loss. We employ an equivalent of a rotating coordinate frame to track the signal bearing-change and use the waveguide invariant theory to compensate the signal range-change by frequency shifting. [Work supported by ONR.]

9:35

3aUW5. Interference suppression by tracking and integrating over source motion. T. C. Yang (Naval Res. Lab., Washington, DC 20375)

In this paper, we will show that the beam-domain motion-compensation algorithm can be used to suppress an interference signal when the interference source is at a different bearing/range, and/or has a different bearing/range rate than the signal source. The motion-compensation algorithm was originally developed to preserve the adaptive processing signal gain for a moving source. We have found that it also suppresses interference (beyond normal adaptive processing) under the above-stated conditions. The initial signal bearing is assumed known based on conventional processing but the range is assumed unknown. (The signal is either detected or its presence is highly suggested.) Signal bearing rate and range rate are assumed unknown and will be searched for by the motion compensation algorithm, which yields in principle a highest beam power at the true signal bearing and range rate. We employ an equivalent of a rotating coordinate frame to track the signal bearing-change and use the waveguide invariant theory to compensate the signal range-change by frequency shifting. The mismatch in the bearing and range rate between the interference and signal suppresses the interference power. [Work supported by ONR.]

9:50

3aUW6. Reduced beamset adaptive matched field processing. Brian Tracey, Srinivas Turaga, and Nigel Lee (MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, btracey@ll.mit.edu)

Matched field processing (MFP) offers the possibility of improved towed array performance at endfire through range/depth discrimination of contacts. One challenge is that arrays with limited vertical aperture can often resolve only a small number of multipath arrivals. This paper explores ways to capture the array resolution by re-parametrizing the set of MFP replicas. A reduced beamset can be created by performing a singular value decomposition on the MFP replica set. Alternatively, clustering techniques can be used to generate MFP cell families, or regions of similar response. These parametrizations are applied to adaptive MFP algorithms to show speed and performance gains. The use of cell families/regions instead of individual MFP cells also provides a framework for increasing the robustness of MFP by defocusing the MFP beamforming operation. The techniques are demonstrated for shallow-water towed array scenarios. [Work sponsored by DARPA-ATO under Air Force Contract No. F19628-00-C-0002. Opinions, interpretations, conclusions, and recommendations are those of the authors and are not necessarily endorsed by the Department of Defense. Approved for Public Release, Distribution Unlimited.]

10:05–10:20 Break

10:20

3aUW7. Adaptive sonar detection performance prediction in an uncertain ocean. Paul Book, Jeffery Krolik (Dept of Elec. & Computer Eng., Duke Univ., Durham, NC 27708-0291), and Shawn Kraut (Queen's Univ., Kingston, ON K7L 3N6, Canada)

This paper addresses the problem of predicting detection performance when the signal wavefront is uncertain and the noise field directionality is unknown. Passive sonar detection in this scenario typically involves robust adaptive beamforming with limited training data. The classical sonar equation, however, assumes the noise field and signal wavefront are known exactly. In this paper, we use the statistics of the generalized likelihood ratio test (GLRT) for the composite hypothesis of a multirank signal in Gaussian noise with unknown covariance matrix to evaluate the detection threshold (DT) as function of ocean uncertainty and number of noise training snapshots. Further, the trade-off between array gain (AG) and detection threshold (DT) is studied as a function of training sample size in a dynamic interference environment. Detection performance of the GLRT is characterized in terms of bounds on the middle 80th percentile of classical passive sonar figure of merit (FOM) and range-of-the-day (RD) over an ensemble of ocean environments. Different classes of environments including downward refracting and upward refracting scenarios are examined with particular attention to the Florida Straits region. Example performance prediction bounds are presented using real horizontal noise field and environmental data. [Work supported by ONR.]

10:35

3aUW8. Mode excision adaptive beamforming for source detection in an uncertain shallow-water waveguide. Vincent E. Premus (MIT Lincoln Lab., 244 Wood St., Lexington, MA 02420, vpremus@ll.mit.edu)

Passive sonar detection is uniquely characterized by the fact that the acoustic clutter distribution is generally confined to the ocean's surface. There is considerable evidence to support the hypothesis that surfaced and submerged sources are well separated in acoustic mode space, and that shallow-water waveguide normal modes are relatively robust to imperfect environmental knowledge. In this work, the use of mode physics is explored for the purpose of identifying an improved adaptive subspace for submerged source detection in the presence of surface interference. The basic premise is to perform adaptive weight computation in a mode subspace that is weakly excited by the submerged source of interest, yet well coupled to the surface interference. The rationale is to excise as much of the target signature as possible from the sample covariance without excessively compromising the measurement of the interference spatial spectrum. This enables more aggressive nulling of the surface clutter spectrum for a given level of signal gain degradation on the submerged source of interest. In this paper, the algorithm for adaptive mode subspace identification will be discussed, and the theoretical performance as a function of imprecise environmental knowledge and array calibration will be examined for a number of different apertures, including vertical line arrays, horizontal line arrays, and volumetric arrays. [Work sponsored in part by

3a WED. AM

DARPA, under Air Force Contract No. F19628-00-C-0002. Opinions, interpretations, conclusions, and recommendations are those of the author and are not necessarily endorsed by the U.S. Air Force.]

10:50

3aUW9. Noise model beamforming. Catherine H. Frazier, Iman W. Schurman, and Bruce K. Newhall (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723-6099, catherine.frazier@jhuapl.edu)

Adaptive beamformers are usually based on the premise that noise is stationary and Gaussian, and hence completely characterized by a covariance matrix. The standard approach then attempts to use measured data to estimate that covariance and form a set of beamformer weights that optimally rejects the estimated noise. This standard method may not be robust to mismatch between real noise and the noise estimate. In particular, low frequency passive sonar noise is usually nonstationary, owing primarily to the motion of the shipping noise sources. We investigate alternative robust noise models that allow for nonstationarity. One such model [W. A. Kuperman and F. Ingenito, J. Acoust. Soc. Am. **67**, 1988–1996 (1980)] assumes a uniform probability distribution for surface noise sources. This produces an optimal beamformer that is robust in uniformly rejecting surface noise, no matter where sources occur on the ocean surface or how they move. The noise performance of this beamformer will be compared to standard adaptive approaches for array configurations proposed by ONR for the Acoustic Observatory Program. The high fidelity simulation model used for noise prediction includes nonstationary source motion and realistic range-dependent multipath propagation effects. [Work supported by DARPA.]

11:05

3aUW10. Nonexhaustive array processing. Peter Gerstoft, Katherine Kim, David Battle, W. A. Kuperman (Marine Physical Lab., Univ. of California San Diego, La Jolla, CA 92093-0238), William Hodgkiss, and Heechun Song (Univ. of California San Diego, La Jolla, CA 92093-0238)

Modern arrays may contain of order a thousand elements and the subsequent beamforming requires searching over frequencies, beam angle, source–receiver range, source depth, array depth, array tilt, and array bow. The search space, easily on the order of a trillion cells, suggests seeking alternative approaches to the traditional exhaustive methods. Here we pursue a nonexhaustive global search strategy combined with a local search for significant computational gain. The approach will be illustrated using computationally demanding simulated data and real towed-array data.

11:20

3aUW11. Matched-field processing with an L-shaped array. Gregory J. Orris, Michael Nicholas, and John S. Perkins (Naval Res. Lab., Washington, DC 20375)

Data was collected on a 64 element L-shaped Array deployed in two at-sea experiments: RDS-1 off the coast of Halifax, Nova Scotia during late September of 1997, and KWIX-98 off Key West, Florida in August of 1998. The L-shaped array was deployed in a configuration with 32 elements in the horizontal and 32 elements in the vertical. Environmental variability during both experiments proved to be extreme at times due to weather and prevailing currents, making processing with so-called "high-resolution" and multi-frequency matched-field processing difficult. It was observed that fully coherent spatial processing using robust linear methods offered only marginal improvement over treating each leg of the array separately and combining the results in an incoherent processor. Possible explanations for these results are explored along with approaches aimed at improving the system performance. [Work supported by ONR.]

11:35

3aUW12. Quantification of deterministic matched-field source localization error in the face of random model inputs. Peter M. Daly and Gerald T. Hebenstreit (SAIC Ocean Systems Operation, 1710 SAIC Dr., M.S. 1-11-15, McLean, VA 22102, peter.m.daly@saic.com)

Deterministic source localization using matched-field processing (MFP) has yielded good results in propagation scenarios where the non-random model parameter input assumption is valid. In many shallow water environments, inputs to acoustic propagation models may be better represented using random distributions rather than fixed quantities. One can estimate the negative effect of random source inputs on deterministic MFP by (1) obtaining a realistic statistical representation of a signal model parameter, then (2) using the mean of the parameter as input to the MFP signal model (the so-called "replica vector"), (3) synthesizing a source signal using multiple realizations of the random parameter, and (4) estimating the source localization error by correlating the synthesized signal vector with the replica vector over a three dimensional space. This approach allows one to quantify deterministic localization error introduced by random model parameters, including sound velocity profile, hydrophone locations, and sediment thickness and speed. [Work supported by DARPA Advanced Technology Office.]

11:50

3aUW13. Blind sound channel deconvolution via artificial time reversal. Karim G. Sabra and David R. Dowling (Dept. of Mech. Eng., Univ. of Michigan, 2019 Lay Auto Lab 2121, Ann Arbor, MI 48109-2121)

Signals that travel through a sound channel are typically distorted when they are received by a remote listener because of interference arising from multiple propagation paths. This presentation introduces a novel passive acoustic technique for blind deconvolution of broadband signals recorded on a vertical array in an unknown multipath sound channel. The technique, called artificial time reversal or ATR, is based on artificially backpropagating the signals measured at the array to their source location using a broadband Green's function synthesized directly from the measurements. The technique exploits generic features of modal propagation in ocean sound channels to infer a phase relationship between the Green's function and the signal. The technique allows sound-channel spread signals to be compressed to their original length. Computational examples employing a single source are presented and compared with results obtained from oceanic measurements made in May 1997 off the west coast of Italy [Hodgkiss *et al.*, J. Acoust. Soc. Am. **105**, 1597 (1999)]. The temporal correlation between an ATR-compressed signal and the original signal approaches 100% in the computational examples and reaches 90% for the oceanic data. Potential extension of this technique to multiple sources emitting simultaneously in the same frequency band is also presented. [Work supported by ONR; oceanic data provided by Dr. Song of SIO.]

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

to be held jointly with the

ANSI-Accredited U.S. Technical Advisory Group (TAG) Meetings for:
ISO/TC 108 Mechanical Vibration and Shock
ISO/TC 108/SC 1 Balancing, including balancing machines
ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring instruments
ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
and
ISO/TC 108/SC 6 Vibration and shock generating systems

R. J. Peppin, Chair S2
5012 Macon Road, Rockville, Maryland 20852

D. J. Evans, Vice Chair S2 and Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108 Mechanical Vibration and Shock and Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring devices
National Institute of Standards and Technology (NIST), 100 Bureau Drive, Stop 8221, Gaithersburg, Maryland 20899-8221

R. Eshleman, Acting Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 1 Balancing, including balancing machines
Vibration Institute, 6262 Kingery Highway, Ste. 212, Willowbrook, Illinois 60514

A. F. Kilcullen, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures
5012 Woods Road, Hedgesville, West Virginia 25427

R. Eshleman, Vice Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 2 and Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machines
Vibration Institute, 6262 Kingery Highway, Ste. 212, Willowbrook, Illinois 60514

G. Booth, Chair of the U.S. Technical Advisory Group (TAG) for ISO/TC 108/SC 6 Vibration and shock generating systems
44 Bristol Street, Branford, Connecticut 06405-4842

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. There will be a report on the interface of S2 activities with those of ISO/TC 108 and its subcommittees including plans for future meetings of ISO/TC and/or its Subcommittees. The Technical Advisory Groups for ISO/TC 108 and the Subcommittees listed above consists of members of S2 and other persons not necessarily members of those Committees. Open discussion of committee reports is encouraged.

Scope of S2: Standards, specifications, methods of measurement and test, and terminology in fields of mechanical vibration and shock, and condition monitoring and diagnostics of machines, but excluding those aspects which pertain to biological safety, tolerance and comfort.

Session 3pAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

Peter F. Worcester, Chair
Scripps Institution of Oceanography, University of California, San Diego, 9500 Gilman Drive, La Jolla, California 92093-0225

Chair's Introduction—12:55

Invited Paper

1:00

3pAO1. The sound of rainfall at sea. Jeffrey A Nystuen (Appl. Phys. Lab., Univ. of Washington, Seattle, WA 98105, nystuen@apl.washington.edu)

Knowledge of the distribution of oceanic rainfall is needed to more fully describe the marine ecosystem and to provide constraints for climate investigations and for process studies of air–sea interaction. Bubbles generated by raindrop splashes produce a loud and distinctive sound underwater. This sound is a signal that allows the detection and measurement of rainfall at sea. Different raindrop sizes have unique splash characteristics, producing distinctive acoustic signatures for at least four raindrop sizes. These signatures form a mathematical basis for the decomposition of the sound field to measure the drop size distribution in the rain. This information allows the acoustic identification of rainfall type and rainfall rate. The acoustic detection and measurement of rainfall at sea is demonstrated using over 100 buoy-months of data from ocean surface moorings in the tropical Pacific Ocean. This technology is now being transferred to sub-surface instrument platforms. By acoustically monitoring rainfall, and other surface processes including wind speed, from below the surface, many of the fouling, physical damage and vandalism problems that affect surface instruments will be avoided. This technology is passive, introducing no acoustic disturbance into the environment and thus poses no potential harm to marine mammals or other forms of life in the ocean.

WEDNESDAY AFTERNOON, 30 APRIL 2003 ROOM 205, 1:55 TO 3:00 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Robin O. Cleveland, Chair
Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215

Chair's Introduction—1:55

Invited Papers

2:00

3pID1. Hot topics in underwater acoustics. John S. Perkins (Naval Res. Lab., Washington, DC 20375, perkins@ghost.nrl.navy.mil)

Underwater acoustics is a "big tent" discipline that draws from, and makes contributions to, many diverse fields such as wave propagation, physical oceanography, signal processing and ocean engineering. In recent years, concern for the environmental effects on marine life has spawned research into both the effects of man-made noise, and the use of active and passive acoustics to monitor biological activity. Special Sessions at this meeting reflect this diversity. This paper will draw on these Special Sessions to present some exciting examples of current research: (1) Parabolic equation methods for wave propagation, (2) robust passive sonar techniques, (3) inverse methods for determining geoacoustic parameters, (4) high-frequency acoustic interaction with the sea floor, and (5) bioacoustic resonance. In signal processing, one of the most active areas is the exploitation of the concept of time-reversal. In ocean engineering, there has been a tremendous increase in the applications for autonomous undersea vehicles. A brief overview of developments in these areas will be included. [Work supported by ONR.]

2:20

3pID2. Synergistic advances in diagnostic and therapeutic medical ultrasound. Frederic L. Lizzi (Riverside Res. Inst., 156 William St., New York, NY 10038)

Significant advances are more fully exploiting ultrasound's potential for noninvasive diagnosis and treatment. Therapeutic systems employ intense focused beams to thermally kill cancer cells in, e.g., prostate; to stop bleeding; and to treat specific diseases (e.g., glaucoma). Diagnostic ultrasound techniques can quantitatively image an increasingly broad spectrum of physical tissue attributes. An exciting aspect of this progress is the emerging synergy between these modalities. Advanced diagnostic techniques may contribute at several stages in therapy. For example, treatment planning for small ocular tumors uses 50-MHz, 3-D ultrasonic images with 0.05-mm resolution. Thermal simulations employ these images to evaluate desired and undesired effects using exposure stategies with specially designed treatment beams. Therapy beam positioning can use diagnostic elastography to sense tissue motion induced by radiation pressure from high-intensity treatment beams. Therapy monitoring can sense lesion formation using elastography motion sensing (to detect the increased stiffness in lesions); harmonic imaging (to sense altered nonlinear properties); and spectrum analysis images (depicting changes in the sizes, concentration, and configuration of sub-resolution structures.) Experience from these applications will greatly expand the knowledge of acoustic phenomena in living tissues and should lead to further advances in medical ultrasound.

2:40

3pID3. ASA education outreach. Uwe J. Hansen (Dept. of Phys., Indiana State Univ., Terre Haute, IN 47809) and E. Carr Everbach (Swarthmore College, Swarthmore, PA 19081-1397)

A number of very successful Hands-on demo sessions for high school students have been a part of regular ASA meetings for some time. In addition, the Education Committee has organized a series of teacher workshops. These workshops are designed to give high school teachers relatively sophisticated tools to enhance their laboratory content. Workshops for teachers in the elementary grades prepare teachers to use music as a vehicle to introduce additional science concepts. Content and methods associated with both workshops will be discussed. Cyberspace outreach by the ASA was accelerated by the establishment of a Home Page Committee, and more recently by the On-Line Education committee, which is creating an educational website. The website provides a fun way for users to access information including acoustics information, history, demos, and links to the Technical Committee's webpages. The ASA has joined other AIP member societies in developing additional mechanisms, including road shows and nightly news spots.

WEDNESDAY AFTERNOON, 30 APRIL 2003 ROOM 102, 1:00 TO 2:30 P.M.

Session 3pNS

Noise, Architectural Acoustics, Speech Communication, Psychological and Physiological Acoustics, Engineering Acoustics and ASA Committee on Standards: Panel Discussion, ASA's Role in Developing a Policy Statement and Outreach for the ANSI Classroom Acoustics Standard

Edward J. Walsh, Cochair
Boys Town National Research Hospital, 555 North 30th Street, Omaha, Nebraska 68131

Louis C. Sutherland, Cochair
27803 Longhill Drive, Rancho Palos Verdes, California 90275-3908

David Lubman, Cochair
14301 Middletown Lane, Westminster, California 92683

Chair's Introduction—1:00

Last year the Acoustical Society of America (ASA) Panel on Public Policy (POPP) was charged with the responsibility for drafting an ASA position statement for the Classroom Acoustics Standard (ANSI S12.60-2002). A draft policy for this Standard was prepared in January 2003 and was submitted to the Executive Council for approval. Both the Panel on Public Policy and the Executive Council expressed the need for a strong statement of affirmation for the Standard, which would allow the ASA to speak with a unified voice and technical accuracy on this matter of societal concern.

One of the principal goals associated with this first POPP initiative is to generate a policy statement that will allow the ASA to extend its outreach to other scientific, engineering and standardization bodies, as well as to the public at large. The content and impact of the policy document will be discussed, along with suggestions for approaches to increase outreach for the Classroom Acoustics Standard.

Session 3pPA

Physical Acoustics: Thermoacoustics

Matthew E. Poese, Chair
Graduate Program in Acoustics, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804

Contributed Papers

1:00

3pPA1. Cascade thermoacoustic engine. D. L. Gardner and G. W. Swift (Los Alamos Natl. Lab., Los Alamos, NM 87545)

Over the past several years, Los Alamos has successfully fabricated and tested traveling wave thermoacoustic heat engines having efficiencies much greater than those of traditional standing wave thermoacoustic engines. Unfortunately, Gedeon streaming effects present in these traveling wave engines has required using fluid flow ''diodes'' having greater complexity than desired. The cascade thermoacoustic engine recently fabricated and tested at Los Alamos uses three stages in series, one standing wave and two traveling wave, yielding a configuration in which Gedeon streaming is eliminated and efficiency can still be high. The cascade engine will be described and experimental data will be presented. [Work supported by the U.S. DOE Office of Science and Praxair, Inc.]

1:15

3pPA2. Sensorless control of a harmonically driven linear reciprocating electrodynamic machine for a thermoacoustic refrigerator. Tony L. Shearer (Northrup Grumman Electron. Systems, Sykesville, MD), Robert W. M. Smith (Penn State Appl. Res. Lab., State College, PA), Heath F. Hofmann (Penn State Dept. of Elec. Eng., University Park, PA), and Steven L. Garrett (Penn State Grad. Prog. in Acoust., University Park, PA)

In electrically driven thermoacoustic refrigerators, the acoustic pressure and actuator displacement are typically monitored while a nominally harmonic drive frequency is adjusted to achieve a given operating point (e.g., maximum electroacoustic conversion efficiency, acoustic resonance, maximum power, etc.) Most often, pressure and displacement are measured directly, using sensors. Using a model of the electrical and mechanical dynamics of a linear electrodynamic machine and resonant assembly, the actuator displacement and pressure at the piston can be accurately estimated via direct measurement of the voltage and current at the machine terminals. This provides a means for sensorless control of the machine. The estimation method and system parameter extraction method will be described, as well as results of comparisons to sensors. Results from tests on a thermoacoustic refrigerator operating at 3-kW acoustic input power using a sensorless control system implemented to maintain an acoustic resonance will be shown, wherein the refrigerator cold temperature was varied over a practical operating range. The control system tracked the acoustic resonance frequency and achieved the desired phase relationship between pressure and displacement to within 3 deg based on the estimation, thus maintaining essentially optimal electroacoustic conversion efficiency during the test. [Work sponsored by ONR.]

1:30

3pPA3. Practical considerations for the design of bellows seals in thermoacoustic refrigerators. Robert W. M. Smith (Penn State Appl. Res. Lab., P.O. Box 30, State College, PA 16804)

In many acoustic sources designed for thermoacoustic refrigerators, the means to provide coupling from a reciprocating electrodynamic machine to the thermoacoustic load has been a flexible metallic seal (bellows). Employing bellows provides for improved efficiency and less stringent assembly tolerances in comparison with clearance seal designs, at the cost of a part that must endure high cycle fatigue. Often the dominant contribution to the stresses in the bellows is associated with deflection. Because the compressional wave speed in the bellows can be small (15 m/s) one must consider the effect of dynamic stress enhancement. For a given bellows geometry and desired end deflection, an optimum bellows length exists which minimizes the deflection stress. Interestingly, the selection of this optimum length also provides some intrinsic reduction in sensitivity of the computed stresses associated with commercial fabrication tolerances in the bellows. It will be shown that with a given bellows geometry, material design stress limit, and a design using the optimum length, results effectively in an end velocity limit on the bellows. In this way, design with conventional bellows can constrain the power available from some commercially available machines. [Work sponsored by ONR.]

1:45

3pPA4. Numerical modeling of inert gas-condensing vapor thermoacoustic engines. W. V. Slaton (Phys. Dept., Eindhoven Univ. of Technol., P.O. Box 513, 5600 MB Eindhoven, The Netherlands), Richard Raspet, and Robert A. Hiller (Univ. of Mississippi, University, MS 38677)

Recent theoretical work by Slaton and Raspet *et al.* describe the acoustic propagation equation [J. Acoust. Soc. Am. **114**, 1414–1422] and the second-order enthalpy and mass transport equations [J. Acoust. Soc. Am. **114**, 1423–1430] for an inert gas-condensing vapor mixture in a porous medium with an imposed temperature gradient. The acoustic propagation and enthalpy transport equations show that the vapor diffusion effects in the mixture are analogous to the heat diffusion effects in the thermoacoustics of inert gases, and that these effects occur in parallel with the heat diffusion effects in the wet system for proper choice of inert gas and vapor. Writing the acoustic propagation equation as two coupled first-order differential equations in terms of the volumetric velocity and acoustic pressure amplitude and utilizing the conservation of enthalpy in the stack allows the system of equations to be solved numerically by interfacing with the well-established thermoacoustic modeling code, DELTAE. Modeling of various thermoacoustic engines utilizing an inert gas-condensing vapor working fluid will be presented. It will be shown how the COP relative to Carnot and the heat pumping power for thermoacoustic refrigerators can be increased significantly by proper choice of gas mixture. [Work supported by ONR.]

2:00

3pPA5. Impedance tube measurements of thermoviscous functions with a condensable working fluid. D. C. Brown, R. A. Hiller, and R. Raspet (Dept. of Phys., Univ. of Mississippi, University, MS 38677)

Theoretical work by Slaton *et al.* [J. Acoust. Soc. Am. **112**, 1414–1422 (2002); **112**, 1423–1430 (2002)] develops the linear theory for thermoacoustics when the working fluid has a component which undergoes condensation and evaporation on the stack during the acoustic cycle. In order to test these ideas an impedance tube is used to measure the acoustic impedance in a closed-end driver-pipe system which contains a sample stack. The impedance is measured on the driver side of the stack from pressure measurements of two closely spaced microphones. This may be compared to the impedance computed from a numerical integration from the closed end of the pipe through the stack, using the known or assumed values of the thermoviscous dissipation functions in the stack. Measurements and the theory for a variety of stack materials and working fluids will be presented.

2:15

3pPA6. An aeroacoustically driven thermoacoustic heat pump. W. V. Slaton and J. C. H. Zeegers (Dept. of Appl. Phys., Eindhoven Univ. of Technol., P.O. Box 513, 5600 MB Eindhoven, The Netherlands)

The mean flow of gas in a pipe past a side branch, closed at the far end, can excite the resonant acoustic modes of the cavity much like blowing across the top of a bottle. This aeroacoustic whistle can excite very high amplitude acoustic waves within the side branch (easily 10% of the mean pressure) at optimal gas flow rates and mean pressures within the main pipe. The aeroacoustic whistle uses no moving parts to convert part of the power in the mean flow into acoustic power. Likewise a thermoacoustic heat pump, utilizing this acoustic power, uses no moving parts to pump heat and establish (or maintain) a temperature difference across a porous medium. This new combination of an aeroacoustic sound source and thermoacoustic heat pump (with suitable thermoelectric elements) is part of an electric power generation feasibility study for natural gas wells. Reliable electrical power generation down-hole to provide electricity for sensors, communications devices or energy storage units is an important research and development goal. Experimental results will be presented that demonstrate the performance of a simple thermoacoustic heat pump when powered by an aeroacoustic sound source. [Work supported by Shell International Exploration and Production B.V.]

WEDNESDAY AFTERNOON, 30 APRIL 2003 ROOM 204, 1:00 TO 3:00 P.M.

Session 3pPP

Psychological and Physiological Acoustics: Binaural Processing and Spatial Hearing

Pavel Zahorik, Chair

Waisman Center, Room 571, University of Wisonsin–Madison, 1500 Highland Avenue, Madison, Wisconsin 53705

Contributed Papers

1:00

3pPP1. Investigating the precedence effect for noise bursts of different bandwidths. Jonas Braasch (Institut für Kommunikationsakustik, Ruhr-Universität Bochum, 44780 Bochum, Germany)

The fact that the precedence effect partly fails for signals with a very narrow bandwidth raises the question of if this phenomenon requires an across-frequency-band interaction. To investigate this, a psychoacoustic experiment was conducted, in which the perceived lateralization of a noise burst (ITD: 300 μs) in the presence of a reflection (ITD: -300 μs) was measured. The parameters in this experiment were the inter-stimulus interval (ISI: 0.0 ms–4.0 ms) and the bandwidth (100 Hz, 400 Hz and 800 Hz). The data of the listeners clearly show that the localization dominance becomes more stable, especially in regard to the dependence on the ISI, when the bandwidth is increased. At the narrowest tested bandwidth, localization dominance could not be observed for some of the listeners, while others perceived at least the sound coming from the side of the direct sound source, although the position of their auditory events varied strongly with the ISI. A signal analysis on the basis of a modified Lindemann algorithm reveals that, in the first case, the precedence effect does not seem to have any significant influence. The signal analysis also indicates that it is not necessary to assume an interaction across the involved frequency bands and that it is sufficient to simply average over the model outputs of those frequency bands.

1:15

3pPP2. Level effects in monaural and interaural intensity discrimination. Mark A. Stellmack, Neal F. Viemeister, and Andrew J. Byrne (Dept. of Psych., Univ. of Minnesota, 75 E. River Rd., Minneapolis, MN 55455)

Discrimination of interaural level differences (ILDs) is often assessed using a two-interval task in which ILD changes between intervals. In such a situation, overall level is usually roved in order to minimize monaural cues. This roving makes it difficult to assess level-dependent effects. In the present experiment, monaural intensity discrimination in a 2IFC task was compared to ILD discrimination in a single-interval task, thus producing an analogous situation in terms of number of "observations" and eliminating the need for an overall level rove. Monaural intensity DLs and ILD discrimination thresholds were measured as a function of level for 4-kHz tones and for broadband noise. The Weber functions ($10 \log \Delta I/I$ vs I in dB) in the monaural and binaural conditions were parallel. For the noise the Weber functions had slopes close to zero (Weber's law); for the tones the slopes were -0.082 (near-miss to Weber's law). Overall, the binaural thresholds showed a small, approximately 2 dB, advantage over the monaural thresholds. The important aspect, however, is that the level effects seen monaurally are also seen binaurally. This suggests that the basic processes responsible for Weber's law and the near-miss occur prior to binaural interaction. [Work supported by NIDCD DC05343 and DC00683.]

1:30

3pPP3. Online estimation of room reverberation time. Rama Ratnam (Beckman Inst., Univ. of Illinois at Urbana—Champaign, Urbana, IL 61801), Douglas L. Jones, Bruce C. Wheeler, and Albert S. Feng (Univ. of Illinois at Urbana—Champaign, Urbana, IL 61801)

The reverberation time (RT) is an important parameter for characterizing the quality of an auditory space. Sounds in reverberant environments are subject to coloration. This affects speech intelligibility and sound localization. State-of-the-art signal processing algorithms for hearing aids are expected to have the ability to evaluate the characteristics of the listening environment and turn on an appropriate processing strategy accordingly. Thus, a method for the characterization of room RT based on passively received microphone signals represents an important enabling technology. Current RT estimators, such as Schroeder's method or regression, depend on a controlled sound source, and thus cannot produce an online, blind RT estimate. Here, we describe a method for estimating RT without prior knowledge of sound sources or room geometry. The diffusive tail of reverberation was modeled as an exponentially damped Gaussian white noise process. The time constant of the decay, which provided a measure of the RT, was estimated using a maximum-likelihood procedure. The estimates were obtained continuously, and an order-statistics filter was used to extract the most likely RT from the accumulated estimates. The procedure was illustrated for connected speech. Results obtained for simulated and real room data are in good agreement with the real RT values.

3p WED. PM

1:45

3pPP4. Reduced order modeling of head related transfer functions for virtual acoustic displays. Joel A. Willhite, Kenneth D. Frampton (Dept. of Mech. Eng., Vanderbilt Univ., Box 1592 Station B, Nashville, TN 37235, joel.a.willhite@vanderbilt.edu), and D. Wesley Grantham (Vanderbilt Univ. Medical Ctr., Nashville, TN 37212)

The purpose of this work is to improve the computational efficiency in acoustic virtual applications by creating and testing reduced order models of the head related transfer functions used in localizing sound sources. State space models of varying order were generated from zero-elevation Head Related Impulse Responses (HRIRs) using Kungs Single Value Decomposition (SVD) technique. The inputs to the models are the desired azimuths of the virtual sound sources (from minus 90 deg to plus 90 deg, in 10 deg increments) and the outputs are the left and right ear impulse responses. Trials were conducted in an anechoic chamber in which subjects were exposed to real sounds that were emitted by individual speakers across a numbered speaker array, phantom sources generated from the original HRIRs, and phantom sound sources generated with the different reduced order state space models. The error in the perceived direction of the phantom sources generated from the reduced order models was compared to errors in localization using the original HRIRs.

2:00

3pPP5. Effects of auralization technique and cross-modal visual stimulus on auditory localization in virtual environments. Paul D. Henderson, Rendell R. Torres, Yasushi Shimizu (Prog. in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, hendep2@rpi.edu), Richard Radke, and Brian Lonsway (Rensselaer Polytechnic Inst., Troy, NY 12180)

The performance of auditory localization in audiovisual environments is affected by both the complexity and congruence of the auditory and visual sensory inputs. Subjects were asked to identify the perceived location of a virtual acoustic source placed at varying positions in the horizontal plane and rendered by either conventional stereophonic reproduction or through wavefront reconstruction by a loudspeaker array. The virtual source was auralized in simulated reflective and anechoic acoustic environments both with no visual stimulus and with projected imagery of varying congruence. The localization precision of subjects presented with complex aural stimuli in the form of simulated reflective environments is reduced relative to anechoic conditions. Additionally, array-based wavefront reconstruction provides a significant increase in overall localization performance over stereophonic rendering, particularly for subjects positioned off the primary axis of the loudspeaker array. Subjects presented with simultaneous aural and visual inputs are able to accurately locate the virtual sound source when a large incongruence angle exists (typically more than 30 degrees). However, in the presence of a smaller angular separation between stimuli, a bias towards the visual stimulus is detectable. The results help determine the requirements for accuracy in auralization and reproduction in the creation of virtual multimedia environments.

2:15

3pPP6. The effect of stimulus bandwidth and subject position on horizontal-plane localization with virtual source images. D. Wesley Grantham, Daniel H. Ashmead, Robert S. Wall (Dept. of Hearing and Speech Sci., Vanderbilt Univ. Medical Ctr., Nashville, TN 37232), Kenneth D. Frampton, and J. Andrew Willhite (Vanderbilt Univ., Nashville, TN 37235)

In an anechoic chamber normal-hearing subjects performed a localization task in the frontal horizontal plane. The stimulus was a 200-ms burst of filtered noise. Within a block of trials, half of the presentations (randomly determined) were ''real''—presented from single loudspeakers—and the other half were ''phantoms''—produced by the simultaneous activation of two loudspeakers at $\pm 30°$ using a virtual source imaging technique [Takeuchi *et al.*, J. Acoust. Soc. Am. **109**, 958–971 (2001)]. Both phantom and real sources spanned the azimuthal range $\pm 80°$. When the stimulus was a 4 kHz low-pass filtered noise, rms error was only slighly higher for phantom ($\overline{D}=7.1°$) than for real ($\overline{D}=5.5°$) sources. For 8 kHz low-pass filtered noise, performance remained about the same for real sources, but increased for phantom sources ($\overline{D}=11.5°$). Data will also be reported for conditions in which the subject's position is systematically varied outside the ''sweet spot.'' Results will be discussed in terms of robustness of the virtual imaging technique to stimulus and position factors and its potential usefulness as a tool for the investigation of human auditory spatial perception in static and dynamic environments. [Work supported by NIDCD.]

2:30

3pPP7. The effect of background noise on the perception of auditory distance. Christopher A. Brown, William M. Whitmer, and William A. Yost (Parmly Hearing Inst., Loyola Univ., 6430 N. Kenmore, Chicago, IL 60626)

Five-ms broadband noise bursts were played and recorded through a KEMAR manikin from 6 different distances in a reverberant room. Distances ranged from 4 to 9 feet, and all recordings were 500 ms in duration. A single loudspeaker sat atop a table, and was moved to each distance prior to recording. During testing, stimuli were presented over headphones to listeners seated in a sound-attenuating chamber. In one condition, the ambient sound of the room was mixed with the stimuli, and served to fill in the silent gaps between stimulus presentations. Previous results have shown that distance judgments improve significantly when the silent gaps are filled with ambient noise. In the other conditions on-line noise, which was filtered to approximate the spectrum of the ambient noise, was added instead. Three levels of attenuation were used, -5 dB, 0 dB, and $+5$ dB, relative to the background noise level. The task was to judge the distance of the sound. A repeated measures design was employed, and listeners ran in only one condition on a given day. Preliminary results suggest that judgments are significantly improved with ambient-room noise, as compared to on-line noise. Additional findings and significance will be discussed. [Work supported by NIH.]

2:45

3pPP8. Auditory and visual distance perception: The proximity-image effect revisited. Pavel Zahorik (Waisman Ctr., Univ. of Wisconsin–Madison, 1500 Highland Ave., Madison, WI 53705, zahorik@waisman.wisc.edu)

The proximity-image effect [M. B. Gardner, J. Acoust. Soc. Am. **43**, 163 (1968)] describes a phenomenon in which the apparent distance of an auditory target is determined by the distance of the nearest plausible visual target rather than by acoustic distance cues. Here this effect is examined using a single visual target (an un-energized loudspeaker) and invisible virtual sound sources. These sources were synthesized from binaural impulse-response measurements at distances ranging from 1 to 5 m (0.25-m steps) in the semi-reverberant room ($7.7\ \text{m} \times 4.2\ \text{m} \times 2.7\ \text{m}$) in which the experiment was conducted. Listeners ($n=11$) were asked whether or not the auditory target appeared to be at the same distance as the visual target. Within a block of trials, the visual target was placed at a fixed distance of 1.5, 3, or 4.5 m, and the auditory target varied randomly from trial-to-trial over the sample of measurement distances. The resulting psychometric functions are consistent with the proximity-image effect, and can be predicted using a simple model of sensory integration and decision in which perceived auditory space is both compressed in distance and has lower resolution than perceived visual space. [Work supported by NIH-NEI.]

Session 3pSP

Signal Processing in Acoustics: Statistical and Random Analysis

Leon H. Sibul, Chair
Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, University Park, Pennsylvania 16804

Contributed Papers

1:30

3pSP1. Statistical estimators of a periodically correlated random process for a voiced speech signal. Lesya B. Chorna (School of Audiol. & Speech-Lang. Pathol., Univ. of Memphis, 807 Jefferson Ave., Memphis, TN 38105, lchorna@memphis.edu)

A stochastic model in the form of a periodically correlated random process (PCRP) was applied to a voiced speech signal. In such a signal, the amplitudes and phases of the harmonic components varied randomly; therefore, a stochastic model of the signal was appropriate. A description of the signal in terms of a stationary random process allowed an analysis of the spectral density of the oscillations, but the phase (temporal) structure remained unconsidered. To reveal this structure, a PCRP was used. The speech signal was processed by the co-phase analysis in the time domain and the component analysis in the spectral domain. The algorithms were based on a specific property of the PCRP that the samples selected with a period of correlation formed a series of related stationary sequences. Each sequence was then analyzed using the theory of stationary processes. This technique was used in a study of 70 recordings of the vowel /a/. The recordings were obtained from people with a normal heart rate and provoked arrhythmia. Statistical estimators of the vocal signal calculated on the basis of the PCRP model were statistically different for the normal and arrhythmic heart rates.

1:45

3pSP2. Fisher information for a Gaussian random variable with unequal real and imaginary covariances: Derivation and application to beamforming. Sandra L. Collier (U.S. Army Res. Lab., AMSRL-CI-EE, 2800 Powder Mill Rd., Adelphi, MD 20783-1197, scollier@arl.army.mil) and D. Keith Wilson (Eng. Res. and Development Ctr., U.S. Army Cold Regions Res. and Eng. Lab., Hanover, NH 03755-1290)

There are many physical situations in which a received signal may be modeled as a complex Gaussian random variable. One such situation occurs when an acoustic wave is strongly scattered as a result of propagation through a random medium such as the atmosphere or ocean. However, there are also many conditions under which this model fails to provide an adequate representation. A specific example is the geometric acoustics regime, in which diffraction and scattering by the medium are both weak and the variance of the phase fluctuations is much larger than the variance of the log-amplitude fluctuations. The reduced wavefunction is the wavefunction of a signal propagating in an inhomogeneous medium normalized by the wavefunction in free space. A statistical model is developed for a reduced wavefunction whose real and imaginary components are Gaussian random variables with unequal covariances. A linear transformation is performed and the probability density function of the received signal at a passive array is calculated. The Fisher information is calculated for special conditions of the transformation that are of interest to acoustic beamforming. The coupling of the Cramer–Rao lower bounds on the parameter estimates (e.g., the angles of arrival, source phase, and medium parameters) is addressed.

2:00

3pSP3. Gibbs sampling for multipath arrival time estimation in an uncertain environment. Michele Picarelli and Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Intitute of Technol., Newark, NJ 07102)

Estimation of time delays and amplitudes of multipath arrivals is of great interest in many fields because of the information that can be extracted from these characteristics. In underwater acoustics, multipath features of sound signals can reveal the location of the sound-generating source and properties of the propagating medium; their accurate estimation is, thus, desirable. In this work, a scheme is developed for the estimation of multipath arrival times and amplitudes in a noisy environment considering an unknown signal-to-noise ratio. The number of arrivals at the receiver is considered unknown as well, since, realistically, it depends on properties that would be uncertain *a priori*. The method is based on the efficient calculation of the number of paths, their amplitudes, and arrival times using Gibbs sampling to calculate *a posteriori* probability distributions. Results from the new method are compared to those of conventional approaches and show the potential of the new technique for efficient time delay and amplitude estimation in an environment with several unknowns. [Work supported by ONR.]

2:15

3pSP4. Audio fine classification using the statistical analysis of acoustic images. Edward Chilton and Ioannis Paraskevas (Univ. of Surrey, EE+IT, CVSSP Group, GU2 7XH, UK, E.Chilton@surrey.ac.uk)

The fine classification of audio utterances is an important problem because the features that have to be extracted need to be very accurate in order to contribute to effective classification. In this paper, results are presented for a fine classification problem: namely the classification of two groups of different kinds of gunshots. The problem of accurate classification can be divided into two parts: (i) feature extraction and (ii) classification. The more effective the feature extraction, the more effectively the classifier will be able to categorize the various audio samples. In this paper, a novel method for the automatic recognition of acoustic utterances is presented using acoustic images as the basis for the feature extraction. The feature extraction process is based on the time-frequency distribution of an acoustic unit. A novel feature extraction technique based on the statistical analysis of the spectrogram Hartley transform (distribution) and Choi–Williams distributions of the data is reported as well as a brief discussion of the classifier used. The image is compressed using a statistical analysis of the acoustic image formed from the time-frequency distributions of acoustic data. The kurtosis, L-moments and entropy of the distributions, as well as the energy, contrast, etc., of the corresponding co-occurrence matrices of the distributions are calculated and then combined into a feature matrix. These appropriate features are then presented to the classifier. Initial results obtained indicate that the method is capable of accurate discrimination for fine classification.

2:30

3pSP5. Localizing a large-dimensional field of sonobuoys. Nicole E. Collison (Defence R&D Canada–Atlantic, 9 Grove St., P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada) and Stan E. Dosso (SEOS, Univ. of Victoria, Victoria, BC V8W 3P6, Canada)

For target localization, multistatic sonar systems require an adequate knowledge of both the source and receiver positions. In this paper, we use a regularized acoustic inversion method on measured direct-arrival times from several impulsive sources to track a freely drifting sonobuoy field. The shallow-water experiment involved 11 sonobuoys within a 6×8 km field, with 6 sources over approximately 70 min. Regularization allows prior information to be built into the inversion, which in this case consists of estimates (with associated uncertainties) of the source and initial sonobuoy drop positions determined from the GPS position of the aircraft at the instant of drop, as well as a model for smooth sonobuoy tracks. Closely spaced sonobuoys move along similar tracks, although there is considerable movement in different directions over the entire field (260–700 m). Positioning uncertainties are estimated using a Monte Carlo appraisal procedure to be approximately 100 m (absolute) and 65 m (relative). Submitted for the Signal Processing Young Presenter Award.

WEDNESDAY AFTERNOON, 30 APRIL 2003 RENAISSANCE EAST BALLROOM, 3:15 TO 5:15 P.M.

Plenary Session, Business Meeting and Awards Ceremony

Richard Stern, President
Acoustical Society of America

Business Meeting

Presentation of Certificates to New Fellows

Richard H. Campbell	Zoi-Heleni Michalopoulou
Laurel H. Carney	William C. Moss
Bruce D. Cornuelle	Philip A. Nelson
Amy M. Donahue	Marshall H. Orr
Hans P. Gottlieb	Jack E. Randorff
George E. Ioup	Jens H. Rindel
Jules S. Jaffe	Robert S. Schlauch
Leon M. Keer	Scott D. Sommerfeldt
Robert M. Keolian	Mario A. Svirsky
Oswald Leroy	Floyd E. Toole
Raymond Lim	David P. Walsh
Hugh J. McDermott	Beverly A. Wright
Colette M. McKay	Sean F. Wu
Ronald L. McKay	Ning Xiang

Announcement of Prize and Grant

Jeffrey A. Nystuen, 2003 Medwin Prize in Acoustical Oceanography

Lori L. Holt, American Speech, Language, Hearing Foundation's Research Grant in Speech Science

Presentation of Awards

Silver Medal in Psychological and Physiological Acoustics to Brian C. J. Moore

Helmholtz–Rayleigh Interdisciplinary Silver Medal to Arthur B. Baggeroer

R. Bruce Lindsay Award to Dani Byrd

Gold Medal to Richard H. Lyon