

Session 4aAA

Architectural Acoustics: Acoustics of Meeting Facilities

Jesse J. Ehnert, Chair

Newcomb & Boyd, 303 Peachtree Center Avenue, Northeast, Suite 525, Atlanta, Georgia 30303-1277

Chair's Introduction—8:00

Invited Papers

8:05

4aAA1. Convention facility acoustical design. David Marsh (Pelton Marsh Kinsella, 1420 W. Mockingbird Ln., Ste. 400, Dallas, TX 75247)

Convention centers usually include a variety of meeting spaces with widely varying acoustical requirements. Exhibit halls with several hundred thousand square feet of contiguous space are common along with divisible ballrooms and smaller meeting rooms. Some even have upscale boardrooms with built-in audio–video capabilities rivaling modern corporate facilities. This paper provides an overview of appropriate acoustical design criteria for typical occupied spaces in a convention center in terms of background noise levels, sound isolation between adjacent spaces, and acceptable reverberation times. Examples are given from real-world projects.

8:35

4aAA2. Engineering challenges of the acoustics of a political convention. Jack E. Randorff (Randorff and Assoc., 11 W. Canyon View Dr., Ransom Canyon, TX 79366)

The acoustical challenges encountered during the 2000 Republican Convention are discussed. The convention has held in Philadelphia, Pennsylvania's First Union Center. This venue is a dual-purpose facility catering to professional basketball and professional ice hockey. The acoustical needs of the delegates and the broadcast audience are discussed. The technical performance requirements of convention sound reinforcement and media network broadcast feed are outlined. The necessary technical and performance trade-offs are enumerated with respect to the physical constraints, schedule requirements, budget limitations, and technical planning committee expectations. The conversion of a major sporting arena to a large-scale meeting room with reverberation times and general room conditions conducive to good listening was a significant undertaking. The site had been chosen for a preliminary screening visit approximately 2 years before. This presentation is a followup to "Acoustics of Political Conventions—A Review," delivered at the Acoustical Society of America 139th Meeting in Atlanta in June 2000, 2 months before the convention in Philadelphia.

9:05

4aAA3. Optimal acoustical design of commercial meeting and teleconference rooms. Angelo Campanella (Campanella Assoc., 3201 Ridgewood Dr., Hilliard, OH 43026)

Along with a quiet NC-25 environment, speech communication among multiple individuals in a conference room, especially for teleconferencing, should be enhanced. Conferees are often located around a table and along the room perimeter. Speech propagation among all is enhanced by a flat ceiling reflector placed as low as practical over the table and the seated participants. This reinforces direct sound with a first-bounce reflection between all parties and microphones. Ceiling perimeter, 50%–75% of the ceiling area, is absorptive. This arrangement works for a speaker-phone and for microphones placed on the ceiling reflector avoiding paper shuffling noise. In the latter case, loudspeakers are placed on the absorbing perimeter acoustical tile. Absorption must be placed on the sidewalls to avoid flutter echoes. Uneven absorber distribution among surfaces is evaluated with the Fitzroy reverberation time [D. Fitzroy, J. Acoust. Soc. Am. **31**, 893–897 (1959)]. It was found that this RT at 500 Hz should be $[1 + \log(V)]/10$ (about 1/2 s) to avoid a "hollow" timbre in purveyed speech. Examples of good and troublesome teleconference rooms are discussed.

9:35

4aAA4. The importance of early reflections for speech intelligibility in rooms. John Bradley and Hiroshi Sato (M27-IRC, Natl. Res. Council, Montreal Rd., Ottawa, ON K1A 0R6, Canada)

New results demonstrate that in rooms for unamplified speech communication early reflections are important for achieving adequate speech intelligibility and that early reflection energy is typically much greater in magnitude than the direct sound energy. Speech intelligibility was measured using a rhyme test in simulated sound fields that included a direct sound with varied early reflections and reverberant sound along with a constant level of ambient noise. The results confirmed that added early reflection energy is equivalent to increased direct sound energy. The combination of direct sound and early reflections increases the effective signal-to-noise ratio and the related speech intelligibility scores for both impaired and nonimpaired listeners. The new results also show, that for common conditions where the direct sound is reduced, it is only possible to understand speech because of the presence of early reflections. Analyses of measured impulse responses in rooms for speech, in terms of a measure of the benefit from early reflections, show that early reflections can increase the effective signal-to-noise ratio by up to 9 dB. The primary goal of room acoustics design should be to maximize early reflection energy to make possible increased effective signal-to-noise ratios.

Session 4aBB

Biomedical Ultrasound/Bioresponse to Vibration: Ultrasound Propagation in and Through Bone

Robin O. Cleveland, Chair

*Aerospace and Mechanical Engineering, Boston University, 110 Cummington Street, Boston, Massachusetts 02215***Chair's Introduction—8:00***Invited Papers***8:05****4aBB1. Experimental studies of the interaction of ultrasound with bone: What have we learned in 50 years?** Patrick Nicholson (Depts. of Health Sci. and Phys., Univ. of Jyväskylä, Finland)

Despite five decades of study, our knowledge of the interaction of ultrasound with bone remains incomplete. On the one hand, ultrasound propagation in cortical bone is relatively well-understood. Given its low porosity, cortical bone can be treated as a homogeneous anisotropic solid for the purposes of low-MHz ultrasound assessment. Consequently, ultrasound velocity measurements have been used very successfully as an *in vitro* tool to determine elastic constants. Developing useful clinical techniques for cortical bone assessment has been more difficult, although approaches such as critical angle reflectometry look promising. Understanding ultrasound propagation in trabecular bone, a highly porous fluid-saturated solid with a complicated microarchitecture, has proved a tougher problem. A wide range of clinical devices are already in use despite the fact that, with some notable exceptions, there have been few rigorous attempts to probe the underlying physics. This situation is being resolved, with recent studies into the validity of Biot's and Schoenberg's theories for porous and stratified media, respectively, and also work looking at the role of scattering and the value of backscatter measurements. The insights emerging from this new phase of research are likely to trigger a new generation of acoustic techniques for bone characterization.

8:30**4aBB2. Ultrasonic scattering models for cancellous bone.** Pascal Laugier, Frederic Padilla, and Frederic Jenson (Laboratoire d'Imagerie Paramétrique, 15 rue de l'École de Médecine, 75006 Paris, France)

In recent years, quantitative ultrasound (QUS) measurements have played a growing role in the assessment and management of osteoporosis. This development is attributable to the current wide availability of ultrasonic equipment measuring acoustic parameters in transmission at several skeletal sites. QUS provides equivalent fracture risk assessment compared to conventional x-ray absorptiometric techniques. It is generally accepted that transmission QUS represents a surrogate marker for bone mineral density. Several investigations are currently being conducted to assess innovative QUS techniques to determine and utilize the full potential of this technology for the benefit of detecting pathological conditions that affect bone strength. Among these developments, reflection and scattering techniques may be useful for the quantitative analysis of bone microarchitecture. We will review recent experimental data and theoretical modeling of ultrasound scattering by cancellous bone. Good agreement between experimental data and predictions using a weak scattering model has been reported by different authors for the magnitude and the frequency dependence of the backscatter coefficient. We also found good agreement between the experimental mean trabecular thickness derived from the 3D microarchitecture analysis and theoretical predictions. These results open interesting prospects for bone microarchitecture characterization from *in vivo* measurements.

8:55**4aBB3. Distinct US modes in trabecular bones: Properties and modeling.** Mariusz Kaczmarek and Michal Pakula (Bydgoszcz Univ., Inst. of Env. Mech. & Appl. Comp. Sci., Chodkiewiczza 30, 85-064 Bydgoszcz, Poland)

Marrow saturated trabecular bones are two-phase composites in which the propagation of ultrasonic waves is influenced by relative motion of phases. In particular, in studies *in vitro* two longitudinal wave modes are observed with specific behavior of attenuation and dispersion. Although the existence of the modes is predicted by the macroscopic model proposed by Biot some of their basic properties cannot be derived from the model [see Hosokawa and Otani (1997)]. Thus numerous modifications of the macroscopic model have been developed including layered structure [see Hughes *et al.* (1999)] or incorporating scattering [see Strelitzki *et al.* (1998)] in order to obtain better agreement with experimental results. Still, however, some important features such as high attenuation of the fast wave, insignificant dispersion and moderate attenuation of the slow wave, for relatively short waves as compared with the characteristic size of pores or trabeculae are not well described. This paper discusses predictions of the proposed microscopic (or exact) two-phase model developed for chosen cellular geometries [see Kaczmarek *et al.* (2002)]. The comparison of theoretical results with our own and other experimental data is performed. Additionally, references to results obtained for model material (phantoms) with structure corresponding to the models adopted are made.

9:20

4aBB4. Dependence of ultrasonic scattering on frequency and microarchitecture in trabecular bone: Theory and experiment. Keith A. Wear (Food and Drug Administration, 12720 Twinbrook Pkwy., Rockville, MD 20857)

Measurements of ultrasonic properties of calcaneus (heel bone) have been shown to be effective for the diagnosis of osteoporosis. However, the mechanisms underlying the interaction between ultrasound and bone are currently not well understood. A model that predicts backscatter from trabecular bone has been developed. Scattering is assumed to originate from the surfaces of trabeculae, which are modeled as long, thin, elastic cylinders with radii small compared with the ultrasonic wavelength. Experimental measurements of backscatter using broadband ultrasound centered at 500 kHz from 43 trabecular bone samples (from human calcaneus) *in vitro* have been performed. Microcomputed tomography has been performed on all 43 samples in order to measure microarchitectural features. The theory correctly predicts the measured dependences of backscatter on ultrasonic frequency and trabecular thickness. [Funding from the FDA Office of Womens Health is gratefully acknowledged.]

9:45

4aBB5. Continuous mapping of the acoustic properties of biological materials with emphasis on bone. Sidney Lees (The Forsyth Inst., Boston, MA 02115, sleeves@forsyth.org)

Instruments determine the time of flight through a prepared section of bone of uniform thickness. A multigated receiver can be set to capture successive pulse echoes. A clock turned on by the first pulse echo is turned off by the next. A faster driven transducer maps the bone section. Several tissue properties can be calculated from the time of flight, the amplitude, thickness of the bone section and its density. In particular, the variation of the elastic properties provides a detailed picture of stress distribution on the bone. Several examples will be shown.

10:10–10:25 Break

10:25

4aBB6. QUS devices for assessment of osteoporosis. Christian Langton (Ctr. for Metabolic Bone Disease, Univ. of Hull and Hull & East Yorkshire Hospitals Trust, Hull HU3 2RW, UK)

The acronym QUS (Quantitative Ultrasound) is now widely used to describe ultrasound assessment of osteoporosis, a disease primarily manifested by fragility fractures of the wrist and hip along with shortening of the spine. There is currently available a plethora of commercial QUS devices, measuring various anatomic sites including the heel, finger, and tibia. Largely through commercial rather than scientific drivers, the parameters reported often differ significantly from the two fundamental parameters of velocity and attenuation. Attenuation at the heel is generally reported as BUA (broadband ultrasound attenuation, the linearly regressed increase in attenuation between 200 and 600 kHz). Velocity derivatives include bone, heel, TOF, and Adv. Further, velocity and BUA parameters may be mathematically combined to provide proprietary parameters including “stiffness” and “QUI.” In terms of clinical utility, the situation is further complicated by ultrasound being inherently dependent upon “bone quality” (e.g., structure) in addition to “bone quantity” (generally expressed as BMD, bone mineral density). Hence the BMD derived WHO criteria for osteoporosis and osteopenia may not be directly applied to QUS. There is therefore an urgent need to understand the fundamental dependence of QUS parameters, to perform calibration and cross-correlation studies of QUS devices, and to define its clinical utility.

Contributed Papers

10:50

4aBB7. On the relationship of ultrasonic properties to density and architecture in trabecular bone. Patrick Nicholson and Mary Bouxsein (Orthopedic Biomechanics Lab., BIDMC and Harvard Med. School, Boston, MA 02115)

As previously reported elsewhere, we have made ultrasonic measurements in human trabecular bone and have explored relationships with microstructural properties, the latter derived from microcomputed tomography. However, multicollinearity in these data means that conventional regression analysis cannot reliably identify the underlying causal relationships. In an effort to move beyond such limitations, we used our experimental data to test some models of possible interactions between ultrasound and bone. In particular, we compared several models for predicting acoustic velocity in two-phase media as a function of the bone volume fraction. We found good agreement only with the theory of Kuster and Toksoz [Geophysics 39, 587 (1974)] based on scattering by an effective medium. Turning our attention to attenuation, we examined relationships with trabecular thickness (Tb.Th) and the number of trabeculae per unit volume (Tb.N). The exponent relating attenuation to Tb.Th was 3.2, lower than the value of 4 predicted for long wavelength inelastic scattering by cylinders. This may be due to multiple scattering, since restricting the analysis to specimens with relatively low Tb.N yielded an exponent of 3.9. The exponent relating attenuation to Tb.N was 1.3, higher than the expected value of unity, which may again reflect the influence of multiple scattering.

11:05

4aBB8. Development and validation of a multiecho computer simulation of ultrasound propagation through cancellous bone. Christian Langton and Luke Church (Ctr. for Metabolic Bone Disease, Univ. of Hull and Hull & East Yorkshire Hospitals Trust, Hull HU3 2RW, UK)

Cancellous bone consists of a porous open-celled framework of trabeculae interspersed with marrow. Although the measurement of broadband ultrasound attenuation (BUA) has been shown to be sensitive to osteoporotic changes, the exact dependence on material and structural parameters has not been elucidated. A 3-D computer simulation of ultrasound propagation through cancellous bone has been developed, based upon simple reflective behavior at the multitude of trabecular/marrow interfaces. A cancellous bone framework is initially described by an array of bone and marrow elements. An ultrasound pulse is launched along each row of the model with partial reflection occurring at each bone/marrow interface. If a reverse direction wave hits an interface, a further forward (echo) wave is created, with phase inversion implemented if appropriate. This process is monitored for each wave within each row. The effective received signal is created by summing the time domain data, thus simulating detection by a phase-sensitive ultrasound transducer, as incorporated in clinical systems. The simulation has been validated on a hexagonal honeycomb design of variable mesh size, first against a commercial computer simulation solution (Wave 2000 Pro), and second, via experimental measurement of physical replicas produced by stereolithography.

4a THU. AM

4aBB9. Acoustic properties of bovine cancellous bone from 0.5 to 1.5 MHz.^{a)} Kang Il Lee, Heui-Seol Roh, and S. W. Yoon (Acoust. Res. Lab., Dept. of Phys., Sung Kyun Kwan Univ., Suwon 440-746, Republic of Korea, swyoon@skku.ac.kr)

Most previous studies using ultrasound for the diagnosis of osteoporosis have employed ultrasound in the frequency range from 0.2 to 1 MHz. Acoustic properties of 14 defatted bovine cancellous bone specimens were investigated *in vitro*. Speed of sound (SOS) and broadband ultrasonic attenuation (BUA) were measured using two matched pairs of transducers with the center frequencies of 1 and 2.25 MHz, respectively, in order to cover a broad frequency range from 0.5 to 1.5 MHz. In this frequency range, SOS and BUA show significant linear positive correlations with apparent bone density. These results suggest that the frequency range from 0.5 to 1.5 MHz may also be useful in the diagnosis of osteoporosis. [Work supported by BK21 Program, KRF (KRF-2000-015-DP0718), and KOSEF (KOSEF-2000-0238-100) in Korea.] ^{a)}For (Biomedical Ultrasound/Bioresponse to Vibration) Best Student Paper Award.

4aBB10. A noninvasive method for focused ultrasound surgery through the human skull. Gregory Clement and Kullervo Hynynen (Dept. of Radiol., Harvard Med. School and Brigham and Women's Hospital, Boston, MA 02115)

A technique for focusing ultrasound through the skull bone is described and verified. The approach is based on a layered wave-vector-frequency domain model, which simulates ultrasound propagation through the skull bone using input from CT scans of the head. The algorithm calculates the driving phase of each element in a transducer array in order to maximize the signal at the intended focus. This approach is tested on ten *ex vivo* human skulls using a 500-element hemispherical array operated at 0.74 MHz. A stereotaxic reference frame is affixed to the skulls in order to provide accurate registration between the CT images and the transducer. The focal quality is assessed with a hydrophone placed inside of the skull. In each trial the phase correction algorithm successfully restored the focus inside the skull in a location within 1 mm from the intended focal point. Focusing at high powers (>800-W electrical input) is demonstrated using a brain phantom placed inside a skull. The results demonstrate the feasibility of using the method for completely noninvasive ultrasound brain surgery and therapy.

THURSDAY MORNING, 6 JUNE 2002

BRIGADE ROOM, 9:00 TO 11:50 A.M.

Session 4aEA

Engineering Acoustics and Underwater Acoustics: Mine Hunting Sonar

Thomas R. Howarth, Chair

NAVSEA Newport, 1176 Howell Street, Newport, Rhode Island 02841

Chair's Introduction—9:00

Invited Papers

9:05

4aEA1. Minehunting sonar system research and development. Brian Ferguson (Maritime Operations Division—Sydney, Defence Sci. and Technol. Organisation, P.O. Box 44, Pyrmont NSW 2009, Australia)

Sea mines have the potential to threaten the freedom of the seas by disrupting maritime trade and restricting the freedom of maneuver of navies. The acoustic detection, localization, and classification of sea mines involves a sequence of operations starting with the transmission of a sonar pulse and ending with an operator interpreting the information on a sonar display. A recent improvement to the process stems from the application of neural networks to the computed aided detection of sea mines. The advent of ultrawideband sonar transducers together with pulse compression techniques offers a thousandfold increase in the bandwidth-time product of conventional minehunting sonar transmissions enabling stealth mines to be detected at longer ranges. These wideband signals also enable mines to be imaged at safe standoff distances by applying tomographic image reconstruction techniques. The coupling of wideband transducer technology with synthetic aperture processing enhances the resolution of side scan sonars in both the cross-track and along-track directions. The principles on which conventional and advanced minehunting sonars are based are reviewed and the results of applying novel sonar signal processing algorithms to high-frequency sonar data collected in Australian waters are presented.

9:35

4aEA2. Current development of high-resolution MCM minehunting sonars for small AUVs. Kerry W. Commander, James T. Christoff, and Joseph L. Lopes (Coastal Systems Station, Code R21, Panama City, FL 32407-7001, commanderkw@nsc.navy.mil)

The U.S. Navy recently awarded contracts to develop a new generation of autonomous underwater vehicles (AUVs) and high-performance acoustic sensors that are compatible with these vehicles. These new AUV systems will place minehunting crews and marine mammals well out of dangerous waters in the future. The requirement that these vehicles be small, durable, and easily launched and recovered places severe constraints on the sensor system payloads. The sensor systems must be rugged, low-powered, and small, yet effective enough to justify the risk to the vehicle. They must also have very low false alarm rates (high clutter rejection) because reacquisition and neutralization are costly, time consuming, and closely related to the number of targets found. These new

acoustic sensor systems will use new broadband techniques along with high-resolution multi-aspect designs to achieve their goals from such small platforms. The acoustic sensor packages will consist of advanced very high-resolution Ahead Looking Sonars (ALS) and very high-resolution Synthetic Aperture sonars (SAS). This paper will review the new sonar system designs being considered by the Navy.

10:05

4aEA3. Multi-platform sonar concepts for buried target detection and classification. Joseph R. Edwards, Te-Chih Liu, and Henrik Schmidt (Dept. of Ocean Eng., MIT, 77 Massachusetts Ave., Rm. 5-435, Cambridge, MA 02139)

Buried target classification is of paramount importance in mine countermeasures (MCM) applications. The Generic Oceanographic Array Technology Sonar (GOATS) project approaches this fundamental problem by exploiting the assets of autonomous underwater vehicles (AUVs). Recent developments in unmanned vehicle technology have opened the door for new sonar design concepts that may be applied to create alternative target detection and classification methods in complex environments. Adaptive vehicle behaviors provide the potential for the sonar system to adjust its geometry for optimal performance. The vehicles can also access very shallow water environments, which allows for more complete multi-aspect views of targets of interest than are possible with remote towed array systems. In this paper, it is demonstrated that the multi-platform system can provide improved detection and classification capability, while at the same time reducing the computational requirements. Data from prior GOATS experiments (1998 and 2000) are used to illustrate the effectiveness of the multi-platform sonar concepts, and real-time simulations are used to show the planned implementation of these techniques on-board the AUVs for GOATS'02.

10:35–10:50 Break

10:50

4aEA4. Toward a robust target scattering measurement sonar for low frequencies. Brian H. Houston, H. J. Simpson, S. Liskey, T. Yoder, and J. A. Bucaro (Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375)

In order to develop sonar systems that are non-image-based and that take advantage of target structural acoustics features, broadband capabilities have to be created to operate in shallow water and in the nontraditional frequency band below 30 kHz. Here, source wavelength–aperture ratios are high, resulting in significant source divergence and multipath. At the lowest frequencies, evanescent wave penetration is important in the detection of structures buried below the sediment–fluid interface. Further, the recovery of scattered energy requires nontraditional vertical apertures and synthetic array techniques that have both similarities and differences when compared to high-frequency imaging-based systems. The differences include relatively large resolution cell sizes, robustness to vehicle motion, and the clear potential for higher coverage rates. Towards this end, we discuss the precise measurement of low-frequency scattering cross section in shallow water using an UUV-based synthetic array approach. Broadband environmental acoustic propagation data taken in a bay environment will be presented, together with synthetic aperture data taken with a 20-m horizontal rail. The results of these measurements and the implications for UUV-based broadband sonars will be discussed.

11:20

4aEA5. Obstacle avoidance sonar for submarines. Albert C. Dugas and Kenneth M. Webman (Naval Undersea Warfare Ctr. Div. Newport, 1176 Howell St., Newport, RI 02841-1708, mankm@npt.nuwc.navy.mil)

The Advanced Mine Detection Sonar (AMDS) system was designed to operate in poor environments with high biological and/or shallow-water boundary conditions. It provides increased capability for active detection of volume, close-tethered, and bottom mines, as well as submarine and surface target active/passive detection for ASW and collision avoidance. It also provides bottom topography mapping capability for precise submarine navigation in uncharted littoral waters. It accomplishes this by using advanced processing techniques with extremely narrow beamwidths. The receive array consists of 36 modules arranged in a 15-ft-diameter semicircle at the bottom of the submarine sonar dome to form a chin-mounted array. Each module consists of 40 piezoelectric rubber elements. The modules provide the necessary signal conditioning to the element data prior to signal transmission (uplink) through the hull. The elements are amplified, filtered, converted to digital signals by an A/D converter, and multiplexed prior to uplink to the inboard receiver. Each module also has a downlink over which it receives synchronization and mode/gain control. Uplink and downlink transmission is done using fiberoptic telemetry. AMDS was installed on the USS Asheville. The high-frequency chin array for Virginia class submarines is based on the Asheville design.

Session 4aMU

Musical Acoustics and Psychological and Physiological Acoustics: Music Recognition Systems I

James W. Beauchamp, Chair

*School of Music, Department of ECE, University of Illinois at Urbana-Champaign, 1114 West Nevada, Urbana, Illinois 61801***Chair's Introduction—8:30***Invited Papers***8:35**

4aMU1. The existence region for melodic pitch and computational models. Roy D. Patterson, Katrin Krumbholz (Ctr. for the Neural Basis of Hearing, Dept. of Physiol., Univ. of Cambridge, Downing St., Cambridge CB2 3EG, UK), and Daniel Pressnitzer (Ircam-CNRS, 75004 Paris, France)

The pitch of a note is closely related to the repetition rate of the acoustic wave, or the spacing between the harmonics in the Fourier spectrum of the wave. However, algorithms that extract the repetition rate or harmonic spacing from the sound itself, without regard for auditory processing, would assign pitch perceptions to notes that musicians would not attempt to use to convey melody. If a transcription algorithm is to correctly reject notes that humans would not use, it must include knowledge of the complex constraints that govern the existence region for melodic pitch. Similarly, composers designing synthetic tonal instruments need to know whether the listener will hear the melody the instrument is intended to convey. Psychophysical studies have recently been used to delimit the region of melodic pitch within the overall domain of pitch perception, and to distinguish melodic pitch from rattle pitch, rumble pitch, and flutter pitch, none of which conveys melody successfully. This paper describes how time-domain computational models of auditory processing explain the existence region of melodic pitch, and how they might be used to improve melody transcription and instrument design. [Work supported by the UK Medical Research Council, G9901257.]

9:00

4aMU2. Fundamental frequency estimation of singing voice. Alain de Cheveigné (CNRS, Ircam, 1 Pl. Igor Stravinsky, 75004 Paris, France, cheveign@ircam.fr) and Nathalie Henrich (CNRS, LAM, 11 Rue de Lourmel, 75015 Paris, France)

A method of fundamental frequency (F_0) estimation recently developed for speech [de Cheveigné and Kawahara, *J. Acoust. Soc. Am.* (to be published)] was applied to singing voice. An electroglottograph signal recorded together with the microphone provided a reference by which estimates could be validated. Using standard parameter settings as for speech, error rates were low despite the wide range of F_0 s (about 100 to 1600 Hz). Most "errors" were due to irregular vibration of the vocal folds, a sharp formant resonance that reduced the waveform to a single harmonic, or fast F_0 changes such as in high-amplitude vibrato. Our database (18 singers from baritone to soprano) included examples of diphonic singing for which melody is carried by variations of the frequency of a narrow formant rather than F_0 . Varying a parameter (ratio of inharmonic to total power) the algorithm could be tuned to follow either frequency. Although the method has not been formally tested on a wide range of instruments, it seems appropriate for musical applications because it is accurate, accepts a wide range of F_0 s, and can be implemented with low latency for interactive applications. [Work supported by the Cognitique programme of the French Ministry of Research and Technology.]

9:25

4aMU3. Separation of musical instruments based on amplitude and frequency comodulation. Barry D. Jacobson (Harvard-MIT Div. of Health Sci. and Technol., Cambridge, MA), Gert Cauwenberghs (Dept. of Elec. and Computer Eng., Johns Hopkins Univ.), and Thomas F. Quatieri (MIT Lincoln Lab.)

In previous work, amplitude comodulation was investigated as a basis for monaural source separation. Amplitude comodulation refers to similarities in amplitude envelopes of individual spectral components emitted by particular types of sources. In many types of musical instruments, amplitudes of all resonant modes rise/fall, and start/stop together during the course of normal playing. We found that under certain well-defined conditions, a mixture of constant frequency, amplitude comodulated sources can unambiguously be decomposed into its constituents on the basis of these similarities. In this work, system performance was improved by relaxing the constant frequency requirement. String instruments, for example, which are normally played with vibrato, are both amplitude and frequency comodulated sources, and could not be properly tracked under the constant frequency assumption upon which our original algorithm was based. Frequency comodulation refers to similarities in frequency variations of individual harmonics emitted by these types of sources. The analytical difficulty is in defining a representation of the source which properly tracks frequency varying components. A simple, fixed filter bank can only track an individual spectral component for the duration in which it is within the passband of one of the filters. Alternatives are therefore explored which are amenable to real-time implementation.

9:50

4aMU4. Separation of harmonic sounds using multipitch analysis and linear models for the overtone series. Tuomas Virtanen and Anssi Klapuri (Signal Processing Lab., Tampere Univ. of Technol., P.O. Box 533, FIN-33101 Tampere, Finland, tuomasv@cs.tut.fi)

A signal processing method for the separation of concurrent harmonic sounds is described. The method is based on a two-stage approach. First, a multiple fundamental frequency estimator is applied to find initial sound parameters which are reliable, but inaccurate and static. Second, time-varying sinusoidal parameters are estimated in an iterative algorithm. The harmonic structure is retained by keeping the frequency ratio of overtones constant over time. Overlapping harmonic components are resolved using linear models for the overtone series. In practice, the models retain the spectral envelope continuity of natural sounds. Simulation experiments were carried out using generated test signals, which were random mixtures of two to six notes from recorded natural instruments. The system is able to produce meaningful results in all polyphonies, the quality of separated sounds gradually degrading along with the polyphony. Some denoising algorithms were applied to suppress nonstationary noise component, such as drums in real-world music signals. However, the usability of the system for real musical signals is still quite limited.

10:15–10:25 Break

10:25

4aMU5. Comparison of machine and human recognition of isolated instrument tones. Ichiro Fujinaga (Peabody Conservatory of Music, Johns Hopkins Univ., Baltimore, MD 21202)

This paper describes three different machine recognition experiments and a recently conducted human experiment in order to compare the abilities of machines and humans to recognize isolated instrument tones. The computer recognition software is based on the Lazy Learning Machine, which is an exemplar-based learning system using a k -nearest neighbor (k -NN) classifier with a genetic algorithm to find the optimal set of weights for the features to improve its performance. The performance of the software was progressively improved by adding more features. These include centroid and other higher order moments, such as skewness and kurtosis, the velocities of moments, spectral irregularity, trispectrum, and time-domain envelope shape. Also, realtime recognition is now possible by using Miller Puckett's PD, a realtime software synthesis system, and his fiddle~ object. The training data was taken from the McGill Master Samples. The human experiment involved eighty-eight conservatory students. Although the average human scores are similar to the machine scores, the best human subjects far exceeded the capabilities of the machine. The excellent performance of the humans in this experiment presents new challenges for timbre-recognition computer models.

10:50

4aMU6. Offline and online tempo detection and beat tracking. Jean Laroche (Creative ATC, 1500 Green Hills Rd., Scotts Valley, CA 95066)

Tempo detection and beat tracking consist of measuring the tempo and estimating beat locations in musical signals. Online beat tracking refers to situations where the tempo at a given instant is estimated with access to a limited portion of the future at this time. In offline beat tracking, the whole track is available for processing, which makes the task somewhat easier. Our algorithms use a short-time Fourier analysis of the signal to extract a time-varying measure of the energy variations in several frequency subbands, resulting in an energy flux signal showing large values around note onsets, percussion hits, and other musically relevant events. At regular intervals (every second), the flux signal is correlated with a series of periodic discrete pulses corresponding to various candidate tempos and beat locations. The tempo- and beat-location candidates showing the best fit are retained for the next stage, where a dynamic programming algorithm is used to select the best series of beat locations, based on a measure of tempo continuity, and consistency between tempo and beat. Dynamic programming is used both for offline and online processing with fairly good results.

11:15

4aMU7. Robust identification/fingerprinting of audio signals using spectral flatness features. Juergen Herre, Eric Allamanche, Oliver Hellmuth, and Thorsten Kastner (Fraunhofer IIS-A, Am Weichselgarten 3, D-91058 Erlangen, Germany, hrr@iis.fhg.de)

Recently, the problem of content-based identification material has received increased attention as an important technique for managing the ever-increasing amount of multimedia assets available to users today. This talk discusses the problem of robust identification of audio signals by comparing them to a known reference ("fingerprint") in the feature domain. Desirable properties of the underlying features include robustness with respect to common signal distortions and compactness of representation. A family of suitable features with favorable properties is described and evaluated for their recognition performance. Some applications of signal identification are discussed, including MPEG-7 Audio.

Session 4aPAa

Physical Acoustics: Thermoacoustics

Ray Scott Wakeland, Chair

Graduate Program in Acoustics, Pennsylvania State University, Applied Science Building,
University Park, Pennsylvania 16804

Contributed Papers

8:00

4aPAa1. Efficiency of acoustic power sources from 200 to 20,000 watts. Philip S. Spoor and John A. Corey (CFIC, Inc., 302 10th St., Troy, NY 12180)

The emergence of acoustic “pulse-tube” refrigerators with no moving parts, and with the capacity to provide tens, hundreds, and even thousands of watts of cooling at cryogenic temperatures, is generating considerable commercial interest. To become economically feasible, these cryocoolers require efficient, compact, and inexpensive sources of acoustic power. The authors will discuss a series of five acoustic drivers available from CFIC that range in power from 200 watts to 20,000 watts, using opposing pairs of STAR™ linear motors. Efficiency data for units throughout the range will be presented, showing the relative importance of different loss mechanisms for each size. In particular we will focus on I^2R losses, which are well understood in the small motors but seem anomalously high in the larger motors.

8:15

4aPAa2. Absence of excess attenuation in Celcor ceramic at low frequencies. Jin Liu, Steven L. Garrett (Grad. Prog. in Acoust., Penn State Univ., University Park, PA 16804, jyl118@psu.edu), Gregory S. Long and Ayusman Sen (Penn State Univ., University Park, PA 16802)

Several studies of the excesses attenuation due to the porosity of Celcor™ material have been published [J. M. Sabatier *et al.*, *J. Acoust. Soc. Am.* **99**, 2430–2432 (1996)]. Since we were planning to use Celcor™ in a refrigeration application, we were concerned with this reported excess loss mechanism. To provide an independent measurement of the loss at our frequencies of interest, we placed a 38 mm long sample of Celcor™ at the center of a 700 mm long electro-dynamically-driven plane wave resonator. Using the even modes, as per the technique described by Moldover *et al.* [*Modern Acoustical Techniques for the Measurement of Mechanical Properties* (Academic, New York, 2001), pp. 395–399], the quality factors were measured between 230 Hz and 1.6 kHz in air at atmospheric pressure on a pristine sample. Pores (but not channels) of that sample were then blocked by a sealing process that involved an initiator, azobisisobutronitrile (AIBN), mixed with the monomer hydroxyethyl methacrylate. There was no observed decrease in attenuation of even modes for the coated sample below 1.0 kHz. The volume exclusion due to pore sealing was inferred by the increase in even mode resonance frequencies. Thermoviscous losses were computed with DELTAE and compared to measured losses. [Work supported in part by ONR and the United Technologies Corporation Professorship Endowment.]

8:30

4aPAa3. Wet wall thermoacoustics in a single pore. Brian Espino and Larry Wilen (Dept. of Phys. and Astron., Ohio Univ., Athens, OH 45701)

We will discuss thermoacoustic measurements in a single circular pore with a mixture of air and methanol vapor as the working gas. The vapor is in coexistence with liquid methanol capillary-condensed in a thin porous material lining the inside wall of the pore. From measurements with no temperature gradient, the diffusio-acoustic function $F(\lambda_D)$ may be obtained. At low frequencies, condensation/evaporation and vapor diffusion is rapid enough to maintain the methanol partial pressure constant at its

saturated equilibrium value. At high frequencies, the methanol partial pressure undergoes adiabatic pressure oscillations along with the air. $F(\lambda_D)$ describes the transition between these behaviors. $\lambda_D = \sqrt{2R/\delta_D}$, where δ_D is the penetration depth for binary diffusion. The measurements probe a range of λ_D from about 1 to 10. With an applied temperature gradient, thermoacoustic effects due to condensation/evaporation combined with vapor transport are detected. Results will be compared to recent theoretical predictions for wet wall thermoacoustic effects [Slaton *et al.*, *J. Acoust. Soc. Am.* **105**, 65–73 (1999) and references cited within]. [Work supported by ONR.]

8:45

4aPAa4. Measurement of Rayleigh streaming in high-amplitude standing waves using laser Doppler anemometry. Michael W. Thompson and Anthony A. Atchley (Grad. Prog. in Acoust., Penn State Univ., 217 Applied Science Bldg., University Park, PA 16802, mwt-ac@psu.edu)

Current theories for Rayleigh streaming in a long, cylindrical pipe predict dc velocities which increase as the square of the amplitude of the first harmonic, and exhibit a parabolic radial profile. The velocity field in such a pipe, driven at one end and rigidly capped at the other, has been measured using LDA (laser Doppler anemometry) techniques. Analysis of the data produced by the LDA system can be complicated by two things: (i) the data are randomly sampled at a frequency below the Nyquist rate, resulting in a large, uneven sample spacing which is unsuitable for standard discrete Fourier transform techniques; and (ii) the dc velocity is typically two to three orders of magnitude smaller than the ac velocity, resulting in a very low signal-to-noise ratio. A robust signal processing algorithm is presented which overcomes these two obstacles, permitting fast and accurate estimates of the magnitudes, phases, and uncertainties of the velocity Fourier coefficients. The measured dc velocities agree well with the theoretical predictions when the tube is driven at low amplitudes, but deviate significantly from the theoretical predictions when the tube is driven at high amplitudes. [Work supported by ONR.]

9:00

4aPAa5. Effect of thermal conductivity on acoustic streaming in a channel. Mark F. Hamilton, Yuri A. Ilinskii, and Evgenia A. Zabolotskaya (Dept. of Mech. Eng., Univ. of Texas, Austin, TX 78712-1063)

Acoustic streaming generated in a viscous fluid with thermal conductivity is considered analytically and numerically. The fluid is confined to a channel formed by two parallel plates. The streaming pattern is similar to that in a viscous fluid without thermal conductivity. Specifically, an acoustic standing wave generates inner and outer streaming vortices that circulate in opposite directions. The outer vortex resembles Rayleigh streaming in a viscous fluid without thermal conductivity, and the inner vortex, confined mostly to the boundary layer region, is due to viscous stresses near the wall. Thermal conductivity is found to increase the streaming velocity in a gas by as much as 20%. Dependence of the viscosity coefficient on temperature is also taken into account, and this effect is found to reduce the streaming velocity. However, the first effect dominates the second, and therefore the net result is larger streaming velocities than in the absence of both effects. An analytical expression for acoustic streaming in both nar-

row and wide channels was derived and used to investigate the two effects individually. The analytical solution is in close agreement with numerical results based on the fully nonlinear equations of motion. [Work supported by ONR.]

9:15

4aPAa6. Analytical solution of the regenerator temperature distribution in a thermoacoustic device with mean flow. Nathan T. Weiland and Ben T. Zinn (Schools of Aerosp. and Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, gte852f@prism.gatech.edu)

Previous work by the authors has identified a potential for improvement in the efficiency of a traveling wave thermoacoustic engine by replacing the hot heat exchanger with a steady flow of hot gas. An essential step in determining the feasibility of such an engine lies in determining how the mean flow affects the temperature distribution in the regenerator, which in turn affects its acoustic driving capability. In the current work, the second order time-averaged thermoacoustic energy equation with mean flow is solved analytically for the mean temperature distribution in the regenerator. The resulting expression accounts for the dependence of the working fluid's thermal conductivity and viscosity on temperature, and results in a nearly exponential temperature profile. The specific effects of mean flow, axial conduction, acoustic velocity amplitude, and gas property variation on the temperature distribution in the regenerator will be explored. In turn, the effect of the temperature profile on regenerator performance will also be reviewed. In addition to their usefulness in the feasibility study, these results can also be applied to assess losses in existing thermoacoustic engines and refrigerators where acoustic streaming results in unwanted net mass flux through the regenerator.

9:30

4aPAa7. Minor-loss coefficients for the exit from heat exchangers in thermoacoustics. Ray Scott Wakeland and Robert M. Keolian (Grad. Prog. in Acoust., Penn State Univ., P.O. Box 30, State College, PA 16804, wakeland@psu.edu)

Heat exchanger minor losses in thermoacoustics are sometimes estimated using $\Delta \dot{E} = -(4/3\pi)(\rho_m A/2)K|u_A|^3$, where ρ_m , u_A , and A are the gas density, peak velocity amplitude, and cross-sectional area, and K is a "minor loss coefficient" [G. W. Swift and D. C. Ward, *J. Thermophys.*

Heat Transfer **10**, 652–662 (1996)]. Because values for K are generally unavailable for oscillating flow, steady-flow results are used, including the formula $K_e = (1 - A_1/A_2)^2$ for sudden increase from cross-sectional area A_1 to a larger area A_2 , such as at the exit from a heat exchanger. This steady-flow result assumes a *uniform* cross-sectional velocity profile. In this presentation, non-uniform, time-dependent velocity profiles for laminar oscillating flow are employed to calculate effective minor-loss exit coefficients, which might represent an improvement over the steady-flow formula for calculating flow losses at the exit from heat exchangers in thermoacoustic devices. Differences from the uniform flow calculation can be significant. For example, for $A_1/A_2=0.66$, a reasonable value for a heat exchanger, this calculation of minor loss is as much as 3.4 times greater than the uniform profile result. [Work supported by ONR and the Pennsylvania Space Grant Consortium.]

9:45

4aPAa8. Minor losses and streaming in thermoacoustic devices. Said Boluriaan and Philip J. Morris (Dept. of Aerosp. Eng., Penn State Univ., 233 Hammond Bldg., University Park, PA 16802)

The efficiency of thermoacoustic devices is limited by streaming and viscous losses. Viscous losses are due to energy dissipation in the vicinity of solid boundaries and minor losses. Minor losses are an additional loss of energy associated with transitions between channels and changes in flow direction. The physics behind minor losses in the high-amplitude oscillatory flows encountered in thermoacoustic devices is not well understood and estimates of its value are often based on results for steady flow. Furthermore, the presence of streaming adds to the complexity in the assessment of minor losses. In the current research, a sudden change in cross-sectional area of a resonator is used as a model to study streaming and minor losses. A time-accurate numerical simulation based on the Navier–Stokes equations is performed to calculate the flow variables in an oscillatory flow inside a resonator with a sudden expansion/contraction. A correlation between the pressure loss due to minor losses and different flow parameters involved in the problem is presented. Comparisons are also made with estimates of minor losses based on a quasisteady flow assumption. [Research supported by ONR.]

THURSDAY MORNING, 6 JUNE 2002

STERLINGS ROOMS 2 AND 3, 10:30 A.M. TO 12:00 NOON

Session 4aPAb

Physical Acoustics: Acoustic-Seismic Coupling

Howard B. Wheeler, Cochair

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

James M. Sabatier, Cochair

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

Contributed Papers

10:30

4aPAb1. Resonant frequency response and surface particle velocity profiles for buried land mines: Theory and experiment. James M. Sabatier (U.S. Army CECOM Night Vision and Electron. Sensors Directorate, Ft. Belvoir, VA 22060), Doru Velea (Planning Systems, Inc., Reston, VA 20191), Ning Xiang, and Roger Waxler (Univ. of Mississippi, University, MS 38677)

Acoustic-to-seismic coupling is currently used to detect buried landmines by measuring a contrast in the particle velocity of the air–soil interface directly above the mine (on-target) and away from the mine

(off-target). Field measurements reveal a resonance phenomenon of the mine–soil system at frequencies around 100 Hz. The resonance frequency and the spatial profile of the on/off-target velocity ratio depend on the type of the mine, depth, and soil characteristics. Experimental results for anti-tank mines will be presented. These results will be compared with predictions by a recently developed model for the scattering of normally incident sound off of a landmine. The model assumes that the mine has a compliant top and the soil is an effective fluid. [Work supported by the U.S. Army Communications-Electronics Command Night Vision and Electronic Sensors Directorate and the U. S. Army Research Office.]

10:45

4aPAb2. Analysis of surface wave formation over a grating surface with bounded beams. Wenhao Zhu (Inst. of Microstructural Sci., Natl. Res. Council, Ottawa, ON K1A 0R6, Canada)

It had been demonstrated previously that homogeneous plane waves could not excite effective surface wave over periodic surface structures at low frequencies. In this work, acoustic surface wave over a periodic surface is predicted under limited source incidence (e.g., Gaussian beam). A comblike grating surface is considered in the analysis, for which a theoretical model for predicting the scattered fields has been developed. Numerical results on scattered surface wave as well as bulk waves are given for different parameters of the incident beam. It is found that at low frequencies ($ka \ll 1$, where a is the grating period and k is the wave number in air), it is the inhomogeneity of the source that results in surface wave propagation in the scattered waves. The analysis is also applicable to impedance grating structures.

11:00

4aPAb3. In situ measurements of the fragipan acoustic to seismic coupling signature. Wheeler Howard and Craig J. Hickey (Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677)

The phenomena of acoustic to seismic (A/S) coupling, observed and studied since the 1950s, has most recently been used to detect shallow buried objects [Sabatier and Xiang, *J. Acoust. Soc. Am.* **105**, 1383 (1999); **106**, 2143 (1999)] and monitor detonation of nuclear weapons [Orcutt, *J. Acoust. Soc. Am.* **105**, 1038 (1999)]. At an air-surface interface airborne acoustic energy is coupled into the ground as seismic energy. The ratio of the seismic and airborne waves constitutes the A/S coupling signature, which is distinctive to the underlying structure. Seismic energy received by a geophone at the interface contains information, via reflected waves, about the underlying subsurface layer, media, and boundaries. Of particular interest in the Mississippi River Valley is the fragipan layer. The fragipan is the layer that directly affects the growth of crops, rate of soil erosion, and rate of water absorption in underlying layers. In this presentation, the A/S coupling signature data taken at an agricultural field station and forward model are discussed.

11:15

4aPAb4. Acoustic-to-seismic coupling variations in cold regions. Donald G. Albert (USA CRREL, 72 Lyme Rd., Hanover, NH 03755, dalbert@crrel.usace.army.mil)

Experiments were conducted to investigate the variations that may occur in acoustic-to-seismic coupling arising from changes in local near-surface conditions. The emphasis of the investigations was on cold regions, where many different surface conditions exist and where conditions may change over a short time period from wind, precipitation, freezing, or thawing. The measurements were conducted by recording blank pistol shots with surface geophones and microphones. Results are presented for

grassland, thin and thick seasonal snow covers, polar firn, thin grounded ice, thick glacier ice, and floating river ice. The ratio of induced ground motion to acoustic pressure ranged from 0.5 to 20 micro-meters per second per Pascal. Often two arrivals were detected on the geophones, a high-speed seismic compressional wave followed by the air wave. [Work funded by the U.S. Army.]

11:30

4aPAb5. Nonlinear acoustic techniques for landmine detection: Experiments and theory, Part II. Murray S. Korman (Dept. of Phys., U.S. Naval Acad., Annapolis, MD 21402) and James M. Sabatier (Univ. of Mississippi, University, MS 38677)

A nonlinear acoustic technique for detecting buried landmines has been suggested by Donskoy [*SPIE Proc.* **3392**, 211 (1998); **3710**, 239 (1999)] and verified in a number of experiments by Sabatier *et al.* [*J. Acoust. Soc. Am.* **110**, 2757, 2758 (2001)]. Airborne sound involving two primary frequencies f_1 and f_2 undergo acoustic-to-seismic coupling which interacts with the compliant mine and soil to generate a difference frequency component that can effect the vibration velocity at the surface. An LDV profile across a buried mine at the difference frequency (chosen at a resonance) showed more relative sensitivity than a linear profile. A layer of soil in a large tube, terminated by an airbacked clamped thin elastic plate, is used to model nonlinear effects including Donskoy's "bouncing" soil-mine interface. The frequency spectrum of modes of the plate's response is investigated vs the amount of soil mass loading. In particular, the lowest mode frequency first decreases with loading and then increases. Linear response curves near resonance compare well with the difference frequency response as one primary is varied. [Work supported by the United States Army Communications-Electronics Command Night Vision and Electronic Sensors Directorate.]

11:45

4aPAb6. A fast effective fluid model for the scattering of normally incident sound off of a buried landmine. Doru Velea (Planning Systems, Inc., 12030 Sunrise Valley Dr., Reston Plaza I, Ste. 400, Reston, VA 20191), Roger Waxler (Univ. of Mississippi, University, MS 38677), and James M. Sabatier (Ft. Belvoir, VA 22060)

Landmines buried in the ground can be found acoustically by insonifying the ground and detecting a contrast between the vibratory motion of the ground surface directly above the mine and away from the mine. A model to predict the near-field scattering of low-frequency (~ 100 Hz) acoustic waves off of a landmine is presented. The mine is assumed to be a rigid cylinder with a compliant top. The ground (soil) is modeled as an effective fluid. Because of the short range of the scattered field at low frequencies it has been possible to replace a full space model with a waveguide model that is straightforward to implement numerically. The predictions of the model will be discussed. [Work supported by the U. S. Army Communications-Electronics Command Night Vision and Electronic Sensors Directorate and the U.S. Army Research Office.]

Session 4aPP

**Psychological and Physiological Acoustics and Architectural Acoustics:
Perceptual Effects in Real and Virtual Rooms**

Barbara G. Shinn-Cunningham, Cochair

Cognitive and Neural Systems, Boston University, 677 Beacon Street, Boston, Massachusetts 02215

Murray R. Hodgson, Cochair

*Occupational Hygiene Program, University of British Columbia, 2206 East Mall, Vancouver,
British Columbia V6T 1Z3, Canada*

Chair's Introduction—9:15

Invited Papers

9:20

4aPP1. Perceptual effects in auralization of virtual rooms. Mendel Kleiner, Pontus Larsson, Daniel Vastfjäll (Chalmers Rm. Acoust. Group, Chalmers Univ. of Tech., SE-41296 Gothenburg, Sweden), and Rendell R. Torres (Prog. in Architectural Acoust., Rensselaer Polytech. Inst., Troy, NY 12180-3590)

By using various types of binaural simulation (or “auralization”) of physical environments, it is now possible to study basic perceptual issues relevant to room acoustics, as well to simulate the acoustic conditions found in concert halls and other auditoria. Binaural simulation of physical spaces in general is also important to virtual reality systems. This presentation will begin with an overview of the issues encountered in the auralization of room and other environments. We will then discuss the influence of various approximations in room modeling, in particular, edge- and surface scattering, on the perceived room response. Finally, we will discuss cross-modal effects, such as the influence of visual cues on the perception of auditory cues, and the influence of cross-modal effects on the judgement of “perceived presence” and the rating of room acoustic quality.

9:40

4aPP2. Effects of room reverberation on spatial release from masking. Richard L. Freyman (Dept. of Commun. Disord., Univ. of Massachusetts, Amherst, MA 01003) and Patrick M. Zurek (Sensimetrics Corp., Somerville, MA 02144)

When signal and masker sources are separated horizontally in an anechoic space, one of the primary benefits is that the acoustic shadow from the head leads to a substantial improvement in signal-to-noise ratio in the high frequencies at one of the ears, relative to conditions in which signal and noise are presented from a common location. This presentation will examine how the improved signal audibility created by head shadow is affected by room reverberation. Through room-simulation models, the effects of three variables will be examined: degree of horizontal separation, distance between the listener and the signal and masker, and degree of room reverberation. This paper will present an evaluation of how well the effects of these variables can be predicted by generalized room acoustics equations incorporating room volume and surface absorption. The reduction in high-frequency head-shadow effects in reverberant environments, combined with a disruption of cues for low-frequency binaural release from masking, severely reduces the benefits of spatially separating signal and noise. Fortunately, due to the precedence effect, the perceived locations of signal and noise are accurate even in a reverberant space. The intelligibility benefits of perceived differences in signal and masker locations will be discussed. [Work supported by NIDCD.]

10:00

4aPP3. Effects of reverberation on speech segregation. John F. Culling, Chaz Yee Toh, and Kathryn I. Hodder (School of Psych., Cardiff Univ., P.O. Box 901, Cardiff CF10 3YG, UK)

Perceptual separation of speech from interfering noise using binaural cues and fundamental frequency (F_0) differences is disrupted by reverberation [Plomp, *Acoustica* **34**, 200–211; Culling *et al.*, *Speech Commun.* **14**, 71–96]. Culling *et al.* found that the effect of F_0 differences on vowel identification was robust in reverberation unless combined with even subtle F_0 modulation. In the current study, speech reception thresholds (SRTs) were measured against a single competing voice. Both voices were either monotonized or normally intonated. Each came from recordings of the same voice, but interfering sentences were feminized (F_0 increased

80%; vocal-tract length reduced 20%). The voices were presented from either the same or from different locations within anechoic and reverberant virtual rooms of Culling *et al.* In anechoic conditions, SRTs were lower when the voices were spatially separated and/or intonated, indicating that intonated speech is more intelligible than monotonous speech. In reverberant conditions ($T_{60}=400$ ms), SRTs were higher, with no differences between the conditions. A follow-up experiment introduced sentences with inverted F_0 contours. While acceptable in quiet, these sentences gave higher SRTs in all conditions. It appears that reverberant conditions leave intonated speech intelligible, but make it unseparable, while monotonous speech remains separable but is unintelligible.

10:20–10:30 Break

10:30

4aPP4. Subjective assessment of listening environments in classrooms. Murray Hodgson and Susan Kennedy (SOEH, Univ. of British Columbia, 3rd Fl., 2206 East Mall, Vancouver, BC V6T 1Z3, Canada, hodgson@mech.ubc.ca)

A questionnaire was developed to evaluate subjective perception of the listening environment by university classroom users. The questionnaire was administered to over 5700 students in 30 randomly selected classrooms. The questionnaire recorded students' perceptions of the listening environment and possible modifying factors. A "perception of listening ease" (PLE) score was generated. Items most associated with a poor listening environment were noise from other students and intermittent noises outside the classroom. Over 50% of students reported frequent difficulty with hearing questions from other students. Poorer PLE score was associated with high background noise from students, an instructor who articulates poorly, poorer classroom lighting and "room air," microphone use, and decreased room volume per student. In large classrooms, hearing-impaired students scored PLE lower. Results indicate that PLE is a reliable and valid measure of classroom users' subjective perceptions of the listening environment, and that optimal classroom acoustical design needs to take into consideration "in-use" conditions, as well as physical characteristics of the empty classroom.

Contributed Papers

10:50

4aPP5. Effects of natural and artificial spatialization cues on segregation. Alain de Cheveigné (CNRS, Ircam, 1 Pl. Igor Stravinsky, 75004 Paris, France, cheveign@ircam.fr), Reinhard Gretzky, Alexis Baskind, and Olivier Warusfel (Ircam, 1 Pl. Igor Stravinsky, 75004 Paris, France)

A series of experiments was performed to better understanding factors that determine the clarity or "transparency" of sound scenes, particularly those created by artificial means. Using a paradigm proposed by C. J. Darwin and R. W. Hukin [J. Exp. Psychol. **25**, 617–629], subjects were presented with a target sentence containing one word (chosen among two) that they had to report. Simultaneously they were presented with a distractor sentence containing a second word, temporally aligned with the first. In the absence of segregation cues, subjects scored 50% correct on average. A higher score indicated that simultaneous segregation and sequential grouping mechanisms were both effective. Stimuli were presented by headphones using individual or dummy-head head-related transfer functions (HRTFs), and various combinations of source positions, room effects, and restitution techniques. [Work supported by the Cognitique Programme of the French Ministry of Research and Technology.]

11:05

4aPP6. The influence of reverberation on spatial release of masking in consonant identification. Sasha Devore, Barbara G. Shinn-Cunningham, Nathaniel I. Durlach, and H. Steven Colburn (Hearing Res. Ctr., Boston Univ., 44 Cummington St., Boston, MA 02215, sashad@mit.edu)

This study investigates how reverberation influences spatial release of masking in consonant identification. HRTFs were measured in two rooms with different reverberation characteristics from sources directly in front of and to the right of the listener at a distance of 1 m. These two sets of room HRTFs and pseudo-anechoic HRTFs (time-windowing out the reverberation) were used to simulate a speech target and speech-shaped masker over headphones in two spatial configurations. In both configurations, the target was directly ahead of the listener; the masker was either directly

ahead or 45° to the right of the listener. Subjects were asked to identify the initial or final consonant (one of 12 stop and fricative consonants) in CVC nonsense syllables for different target-to-masker levels. Each spatial configuration and room condition was tested using binaural, monaural-left, and monaural-right headphone presentations. Consonant confusion matrices are analyzed to determine how specific acoustic features of consonants in different frequency regions are affected by reverberation and noise. Comparisons across the different conditions reveal how binaural and spatial cues influence consonant identification and how different acoustic environments affect both monaural and binaural performance. [Work supported in part by AFOSR Grant Nos. F49620-01-1-0005 and F49620-98-1-0442.]

11:20

4aPP7. Effect of reverberation on speech intelligibility in young children. Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, Litovsky@waisman.wisc.edu)

Children spend a majority of their time in noisy and reverberant environments, where a listener's abilities to understand speech and localize its source are compromised. The present study utilizes a novel test, which was developed in our lab for the purpose of assessing the ability of young children to understand speech in the presence of competing sources, and the extent to which they benefit from spatial separation between target speech and competitors. A 4AFC identification procedure was used, with (familiar) words from the Spondee List for targets. Testing was conducted in three rooms with different spatial dimensions and acoustic characteristics (reverberation times T_{60} equal to 250 ms, 500 ms, and 1200 ms). The target was always presented from the front. Conditions were: quiet, with competing sounds near the target, or with competing sounds 90° away from the target. Data were collected from children with normal hearing, ages 4–7 years. Results suggest that, while children benefit from spatial separation of target and competitors, this benefit becomes increasingly smaller as the amount of reverberation in the room increases. This finding may be relevant in our efforts to improve room acoustics and enhance children's ability to learn in classrooms.

Session 4aSA**Structural Acoustics and Vibration: Flow-Induced Vibration and Noise**

Yan-Fan Hwang, Chair

*Applied Research Laboratory, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16804***Chair's Introduction—8:00*****Invited Papers*****8:05****4aSA1. Prediction of flow induced sound: Past, present, and future.** William Blake and Irek Zawadzki (David Taylor Model Basin, MacArthur Blvd., Bethesda, MD 20817)

The prediction of flow-induced vibration and sound has developed into a refined analysis technology. At its genesis the capability to provide engineering evaluations of flow-driven surfaces was crude and semi-empirical. The ability to provide acoustic estimations of fidelity depended on empiricism, similitude, and engineering experience. Very little was known of the physics of flow-structure interaction to permit otherwise. Currently, capabilities to make high-fidelity engineering predictions have benefitted from continual advances in three major areas. Our knowledge of the physics of flow sources has matured to understand acoustically-relevant flow structures and the interaction of flows with surfaces. Our ability to computationally model structural response and acoustic Green's functions has evolved because of parallel advances in structural acoustics. Computational fluid dynamics has developed into a refined tool for simulating flow over complex geometries. It is continuing to evolve as we learn how to model acoustically relevant subsonic flow structures with large eddy simulation and direct numerical simulation. Once principally a technology available only to military application, these tools are becoming more commonplace in industrial applications. This lecture will trace some of these developments, show some examples, and indicate promising areas of advancement.

8:30**4aSA2. Trailing edge noise reduction in a backward-curved impeller.** Gerald C. Lauchle (Grad. Prog. in Acoust., Penn State Univ., 218-B Applied Science Bldg., University Park, PA 16802)

Motorized impellers are used in many air-moving applications including room circulation, duct flow, roof and wall exhaust, and cooling of electronic components in cabinets. These fans are backward-curved centrifugal blowers that operate with no volute casing. These fans radiate broadband noise due to turbulence ingestion and trailing edge (TE) noise generating mechanisms. Considered here are trailing edge noise generation and its reduction in a typical motorized impeller. The sound power of the subject fans is measured in an acoustically transparent test plenum according to ANSI Standard S12.11-1987. Two different serrated TE treatments are designed. The designs assume that a turbulent boundary layer exists at the blade TE, but the actual fan Reynolds number based on chord length is transitional. Therefore, to assure that a turbulent boundary layer exists at the TE, two different inlet turbulators are implemented. These trip the blade boundary layer to a turbulent state. Reported are the effects of the TE serrations and turbulators acting individually on the fan noise, along with the synergistic effects of using them in combinations. Up to 6 dBA of noise reduction is observed when the two are used together. [Work supported by Nortel Networks.]

8:55**4aSA3. Predicting wall pressure fluctuations beneath a turbulent flow.** Theodore M. Farabee (Naval Surface Warfare Ctr., Carderock Div. Signatures Directorate, 9500 MacArthur Blvd., West Bethesda, MD 20817-5700)

The source of excitation for a wide range of flow-induced vibration and noise problems is the unsteady surface pressure field imposed by flow of a turbulent boundary layer (TBL). In addition, for flow-over discontinuities, the TBL surface pressures constitute an incident wave field that is scattered at discontinuities resulting in sound production for even a rigid surface. For these reasons alone, interest in predicting TBL wall-pressure fluctuations has existed as long as there has been an interest in understanding and estimating flow-induced noise and vibration. Current approaches for predicting, or estimating, wall-pressure fluctuations fall into one of three general categories: direct numerical computations, in which the governing equations are solved directly; a statistical modeling approach which couples time-averaged flow calculations to theoretically derived modeling functions for wall pressures; and empirical methods which assume that wall pressures follow a known scaling relationship formed using relatively standard flow parameters that are easy to prescribe for the flow of interest. The merits of using each of these methods for the purposes of assessing flow-induced noise and vibration will be reviewed and discussed.

9:20

4aSA4. Estimation of broadband power radiated from ribbed plates under turbulent boundary layer excitation. M. L. Rumerman (Signatures Directorate, Naval Surface Warfare Ctr. Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817-5700)

“Exact” calculation of the acoustic power radiated due to the vibration of ribbed underwater panels driven by a turbulent boundary layer (TBL) requires consideration of the interactions among ribs and end supports. This paper shows that the power radiated in broad frequency bands may be directly estimated by considering the discontinuities to act independently. Analysis of a string loaded with identical masses or springs, and subjected to a wave-number-white forcing spectral density, is first used to demonstrate that the mean-squared attachment force within a broad frequency band may be estimated by ignoring coupling among attachments. A water-loaded membrane having parallel line mass attachments is then used to show that each attachment can be considered to radiate independently of the others when the acoustic wavelength is less than the spacing between them. These simple models are generalized and combined to develop an estimate of the vibration-related power radiated by finite ribbed steel plate sections in water driven by TBL-like surface pressures. The approximation is compared to results based on an exact formulation for a doubly periodic plate that represents ribs and end supports, and found to agree within 3 dB. [Work supported by ONR.]

9:45

4aSA5. Unit-cell finite element analysis of the response of complex structures to flow excitation. Nathan C. Martin II (BBN Technologies, 10 Fawcett St., Cambridge, MA 02138, nmartin@bbn.com) and Robert N. Dees (BBN Technologies, Old Mystic Mill, 11 Main St., Mystic, CT 06355)

Analytic estimation of the vibration response of rib-reinforced structures excited by turbulent boundary layer (TBL) wall pressure fluctuations is a subject of ongoing interest for numerous applications in airborne and underwater acoustics. Previous investigators have developed a variety of closed form and approximate analytic methods for estimating the response of structures of interest. Fluid-loading and rib-reinforcement effects make problems involving the response of sea-going vessels particularly challenging. Recent advances in finite element based methods and the enhanced capabilities of modern computer hardware now permit the application of the FEA method to such flow-induced vibration problems. This paper describes the application of the unit-cell capability in BBN’s SARA-2D finite element code to evaluate the response of fluid-loaded, rib-reinforced cylinders to TBL excitation. An overview of the unit-cell modeling method is presented, and example results are provided to illustrate the effects of variations in rib and shell dimensions.

10:10–10:25 Break

10:25

4aSA6. On sound generation mechanism by a centrifugal blower. Sean Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202)

Centrifugal blower noise has often been modeled as dipoles and quadrupoles to account for the effects of fluid–structure interaction and turbulence as the impeller rotates. However, many experimental results have shown that sound powers from centrifugal blowers increase with speed to the power of 4–6, which implies the existence of a monopole [Lighthill, Proc. R. Soc., Ser. A **222**, 564–587 (1952)]. This paper demonstrates that such a monopole indeed exists for a blower running inside a heating ventilation and air-conditioning (HVAC) unit of a vehicle. Tests indicate that this monopole is producible by a volumetric fluctuation due to an unsteady rotor. When the blower is operated at the voltage power input specified by the car manufacturer, the sound power increases with the speed to the power of 4. When the blower is installed on a stable shaft and running at the same voltage power input, the volumetric fluctuations are significantly reduced and the sound power increases with speed to the power of 6. This implies that the monopole sound has been effectively replaced by the dipole sound. Since dipole is less effective in generating sound at low speeds than monopole, eliminating rotor fluctuations can lower vehicle HVAC noise levels.

Contributed Papers

10:50

4aSA7. Excitation by flow over an obstructed opening. Paul J. Zoccola, Jr. (Carderock Div. Naval Surface Warfare Ctr., 9500 MacArthur Blvd., West Bethesda, MD 20817, ZoccolaPJ@nswccd.navy.mil)

The effect on flow-induced cavity resonance of the presence of an obstruction in the cavity opening is considered. The presence of a single obstruction or of a grid generally alters the flow so that the excitation occurs on the smaller length scale created by the obstructions. However, discussion of resonant excitation on the length scale encompassing the obstructions has not been found in the literature. For this study, measurements of cavity pressure due to flow over a cavity with an obstruction of varying dimensions in the opening were made. Measurements of the flow field around the single obstruction were also made. The cavity pressure measurements show that flow over an obstructed opening does result in the occurrence of classical resonant excitation at the large length scale. The frequency of the excitation and the amplitude of the response at the large length scale are reduced, depending on the dimensions of the ob-

struction. Flow field results show the effects that an obstruction has on the flow, including effects on the vortex convection velocity and the energy production distribution.

11:05

4aSA8. Vibrations of infinite plates and the mean value vibrations of finite plates excited by turbulent boundary layer flows. Stephen Hambric and Yun-Fan Hwang (ARL/Penn State Univ., P.O. Box 30, State College, PA 16804, sah@wt.arl.psu.edu)

The vibration response of a turbulent boundary layer (TBL) excited flat plate is analyzed using a finite-element model and infinite plate theory. For the finite-element models, discretization sufficient to resolve the convective fluctuations in the flow excitation field is used for the study. Clamped boundary conditions are assumed for the finite-element models and analyses are conducted at a variety of flow speeds and structural loss factors. Two equivalent TBL wall pressure excitation models are applied to the plates: (1) a modified Corcos cross-spectrum model for the finite-element models, and (2) a wave-vector-frequency spectrum model (the

Fourier transform of the modified Corcos cross spectra) for the infinite plate theory. The TBL wall pressure autospectrum is approximated using the model derived by Smolyakov and Tkachenko. The infinite plate predicts the mean value response of the finite plate very well for all speeds and loss factors considered, showing consistency with Skudrzyk's mean value theory and its usefulness for generating sound and vibration estimates.

11:20

4aSA9. Turbulent boundary layer pressure fluctuations at large scales. Wendy Sanders, Carolyn Judge, Eric Winkel, Steven Ceccio (Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, wendusa@engin.umich.edu), David Dowling, and Marc Perlin (Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109)

The pressure fluctuations underneath a turbulent boundary are an important excitation source for noise and vibration for both aircraft and ships. This presentation describes experimental results from a new study of flat-plate turbulent boundary layer pressure fluctuations in water at large scales and high Reynolds number. The experiments were performed at the U.S. Navy's William B. Morgan Large Cavitation Channel in Memphis, TN on a polished flat plate 3.05 m wide, 12.8 m long, and 0.18 m thick. Flow velocity, skin friction, surface pressure, and plate acceleration measurements were made at multiple downstream locations at flow speeds ranging from 0.5 m/s to 19 m/s for a Reynolds number (based on downstream distance) range of several million to 200 million. Dynamic surface pressures were recorded with 16 flush mounted pressure transducers forming an L-array with streamwise dimension of 0.264 m and cross-stream dimension of 0.391 m. Measured 99% boundary-layer thicknesses were typically of order 0.10 m. Results for spatial and temporal correlation functions, as well as auto- and cross-spectra are presented and compared with prior lower-Reynolds-number results using either inner or outer variable scaling. [Work sponsored by the Defense Advanced Research Projects Agency and ONR Code 333.]

11:35

4aSA10. Flow noise predictions of a submerged cylinder under turbulent boundary layer excitations. Kuangcheng Wu (Dept. of Signatures and Hydrodynamics, Northrop Grumman Newport News, 4101 Washington Ave., Newport News, VA 23607, wu_k@nns.com) and Nickolas Vlahopoulos (Univ. of Michigan, Ann Arbor, MI 48109)

The unsteady fluctuated pressure underneath turbulent boundary layers (TBL) is one of major noise sources in moving vehicles. Recently, discretized TBL forcing functions have been applied to planar structures in air [Y. F. Hwang and S. A. Hambric, Noise-Con, 2000; M. Allen and N. Vlahopoulos, Computers and Structures, 2000; M. Allen and N. Vlahopoulos, *Finite Elements in Analysis and Design*, 2001; M. Allen, R. Sbragio, and N. Vlahopoulos, AIAA J. 2001]. This paper discusses prediction of the flow-induced radiated noise and surface responses of a submerged hemisphere-capped cylindrical shell ($L/D=11$). The FEM/IFEM (infinite finite element method) approach is used to calculate structural acoustic transfer functions and to accurately account for the fluid loading effects. The effect on TBL due to the curvature of a cylinder is captured by utilizing the potential flow—boundary layer theory to determine key boundary layer parameters. Predictions of the surface intensity and far field responses are developed through stochastic analysis due to the natural of the TBL excitations. A MATLAB script is generated to determine the power spectral density of the responses. [Work supported by ONR Code 334.]

11:50

4aSA11. Hydrofoil near-wake sound sources at high Reynolds number. Dwayne A. Bourgoyne, Joshua M. Hamel, Carolyn Q. Judge (Univ. of Michigan, Mech. Eng. Rm. 2010, WE Lay Autolab, 1231 Beal Ave., Ann Arbor, MI 48109, dbourgoy@umich.edu), Steve L. Ceccio, and David R. Dowling (Univ. of Michigan, Ann Arbor, MI 48109)

An important hydroacoustic noise source from a fully submerged non-cavitating hydrofoil is often the unsteady separated turbulent flow near its trailing edge. Here, hydroacoustic noise may be produced by boundary layer turbulence swept past and scattered from the foils trailing edge, and

by coherent vortices formed in the foils near-wake. Such vortices may generate an energetic tonal component that rises above the broadband trailing-edge hydroacoustic noise. This presentation describes results of an experimental effort to identify and measure vortical flow features in the near-wake of a two-dimensional hydrofoil at chord-based Reynolds numbers ranging from 0.5 to 60 million. The experiments were conducted at the U.S. Navy's William B. Morgan Large Cavitation Channel with a test-section-spanning hydrofoil (2.1 m chord, 3.0 m span) at flow speeds from 0.25 to 18.3 m/s. Two trailing-edge shapes were investigated, and foil-internal accelerometers were used to monitor structural vibration. Velocity fluctuation spectra were measured in the foils near-wake with a two-component LDV system, and dynamic surface pressures were measured near the foils trailing edge with flush-mounted transducer arrays. Both indicate Reynolds number and trailing-edge shape-dependent vortex shedding. [Significant assistance provided by personnel from NSWC-CD. Work sponsored by Code 333 of ONR.]

12:05

4aSA12. Two structural intensity prediction methods in plates excited by turbulent boundary layers. Michael J. Daley and Stephen A. Hambric (ARL and The Grad. Prog. in Acoust., Penn State Univ., 217 Applied Science Blvd., University Park, PA 16802, mdaley@sabine.acs.psu.edu)

Structural intensity (S-I) fields may be used to identify energy flow paths through a vibrating structure, as well as energy source and sink regions. Boundary layer excitation of structures occurs in numerous aerospace and underwater applications. This study describes two methods of predicting S-I fields in structures excited by turbulent boundary layers. The first prediction method combines well known multiple-input/multiple-output system theory with finite differencing techniques to obtain S-I predictions. The second method uses an analytic computation technique. This technique combines response matrices between applied pressures and response stresses and velocities to form cross spectra matrices with units of power. These cross spectra matrices are proportional to S-I. The two prediction methods are described in detail and a set of case studies incorporating these methods is presented. The case studies are based on a simply supported plate with an attached point damper. The Corcos model of the boundary layer spatial wall pressure correlation is applied to the plate due to its simplicity. The case studies include intensity fields due to flows above, at, and below aerodynamic coincidence (the frequency at which a structural wavelength equals a convective wavelength). [Work supported by The Applied Research Laboratory, Penn State.]

12:20

4aSA13. Effect of mean flow on the analysis of noise and vibration in hydraulic lines. Mardi C. Hastings (Biomed. Eng. Ctr., Ohio State Univ., 1080 Carmack Rd., Columbus, OH 43210)

Recent reports in the literature conclude that convective terms due to mean flow should be included in the analysis to predict noise and vibration in steel hydraulic lines containing compliant hoses [M. L. Munjal and P. T. Thawani, Noise Control Eng. J. **45**, 235–242 (1997)]. In this study analytical predictions for noise and vibration were compared with experimental data obtained from a transmission line excited by a 10-vane pump source. The predictions were based on a transfer matrix analysis that included the fluid–structure interaction but neglected the convective terms associated with mean flow [M. C. Hastings and C.-C. Chen, J. Passenger Cars, SAE Trans. **102**, 1762–1767 (1994)]. Experimental data and analytical predictions were in excellent agreement. These results indicate that mean flow convective terms are negligible and do not need to be included in the analysis of noise and vibration occurring in relatively low-power hydraulic assist systems (such as automotive power steering). A comparison of analytical results with and without mean flow convective terms for higher-power hydraulic systems indicates when the convective terms are needed for an accurate prediction of noise and vibration.

Session 4aSC

Speech Communication: Speech Perception of Normal and Impaired Hearing (Poster Session)

Peggy B. Nelson, Chair

Communication Disorders, University of Minnesota, 115 Shevlin Hall, 164 Pillsbury Drive, Southeast,
Minneapolis, Minnesota 55455

Contributed Papers

All posters will be on display from 8:30 a.m. to 5:00 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:30 a.m. to 10:15 a.m. and contributors of even-numbered papers will be at their posters from 10:15 a.m. to 12:00 noon. To allow for extended viewing time, posters will remain on display until 5:00 p.m.

4aSC1. Speech recognition in continuous and fluctuating maskers by listeners with normal and impaired hearing: Effects of presentation level. Van Summers and Michelle Molis (Army Audiol. & Speech Ctr., Walter Reed Army Medical Ctr., 6900 Georgia Ave. NW, Washington, DC 20307-5001)

Listeners with normal hearing show better speech recognition in the presence of fluctuating background sounds, such as a single competing voice, than in unmodulated noise at the same overall level. These performance differences are greatly reduced in listeners with hearing impairment who generally show little benefit from fluctuations in masker envelopes. If this lack of benefit is entirely due to elevated quiet thresholds and the resulting inaudibility of low-amplitude portions of signal+masker, hearing-impaired listeners should show an increasing benefit from masker fluctuations as the presentation levels increase. Normally hearing and hearing-impaired listeners were tested for sentence recognition at moderate and high presentation levels in a competing speech-shaped noise, in competing speech by a single talker, and in competing time-reversed speech by the same talker. Normal-hearing listeners showed more accurate recognition in the fluctuating maskers than in unmodulated noise with some evidence that modulated:unmodulated performance differences may decrease at high presentation levels. Hearing-impaired listeners showed more similar performance across maskers and presentation levels. The results suggest that audibility does not completely account for the group differences in performance with fluctuating maskers: supra-threshold processing differences between groups may also contribute to the disparity in performance. [Work supported by NIH.]

4aSC2. The time course of learning during a vowel discrimination task by hearing-impaired and masked normal-hearing listeners. Carrie Davis, Diane Kewley-Port, and Maureen Coughlin (Dept. of Speech & Hearing Sci., Indiana Univ., Bloomington, 200 S. Jordan Ave., Bloomington, IN 47405)

Vowel discrimination was compared between a group of young, well-trained listeners with mild-to-moderate sensorineural hearing impairment (YHI), and a matched group of normal hearing, noise-masked listeners (YNH). Unexpectedly, discrimination of $F1$ and $F2$ in the YHI listeners was equal to or better than that observed in YNH listeners in three conditions of similar audibility [Davis *et al.*, J. Acoust. Soc. Am. **109**, 2501 (2001)]. However, in the same time interval, the YHI subjects completed an average of 55% more blocks of testing than the YNH group. New analyses were undertaken to examine the time course of learning during the vowel discrimination task, to determine whether performance was affected by number of trials. Learning curves for a set of vowels in the $F1$ and $F2$ regions showed no significant differences between the YHI and YNH listeners. Thus while the YHI subjects completed more trials overall, they achieved a level of discrimination similar to that of their normal-

hearing peers within the same number of blocks. Implications of discrimination performance in relation to hearing status and listening strategies will be discussed. [Work supported by NIHDCD-02229.]

4aSC3. Optimal sweep cycle for time-varying comb filters for binaural dichotic presentation to improve speech perception in sensorineural hearing impairment. Alice N. Cheeran, Prem C. Pandey, and Dakshayani S. Jangamashetti (EE Dept., IIT Bombay, Powai Mumbai-400 076, India, pcpandey@ee.iitb.ac.in)

In a previous investigation [P. C. Pandey *et al.*, J. Acoust. Soc. Am. **110**, 2705 (2001)], a scheme using binaural dichotic presentation was devised for simultaneously reducing the effect of increased temporal and spectral masking in bilateral sensorineural hearing impairment. Speech was processed by a pair of time-varying comb filters with passbands corresponding to cyclically swept auditory critical bands, with the objective that spectral components in neighboring critical bands do not mask each other and sweeping of filter passbands provides relaxation time to the sensory cells on the basilar membrane. Presently investigation is carried out to find the optimal value of the sweep cycle. Comb filters used were 256-coefficient linear phase filters, with transition crossovers adjusted for low perceived spectral distortion, 1 dB passband ripple, 30 dB stopband attenuation, and 78–117 Hz transition width. Acoustic stimuli consisted of swept sine wave and running speech from a male and a female speaker. Bilateral loss was simulated by adding broadband noise with constant short-time SNR. Listening tests with stimuli processed using sweep cycles of 10, 20, 40, 50, 60, 80, 100 ms indicated highest perceptual quality ranking for sweep cycle in the 40–60 ms range, with a peak at 50 ms.

4aSC4. Preliminary observations of infants' detection of backward masked tones. Lynne A. Werner and Heather K. Parrish (Dept. of Speech and Hearing Sci., Univ. of Washington, 1417 NE 42nd St., Seattle, WA 98105-6246, lawerner@u.washington.edu)

Backward masked thresholds appear to have a prolonged developmental course compared to other measures of auditory capacity, but backward masking has not been studied in infant listeners. The present study examined 7–9-month-old infants' detection of 20-ms, 1-kHz pure tones masked by a 50-ms, 2.5-kHz-lowpass noise. The interval between the offset of the tone and the onset of the masker was 0 ms. Both stimuli had 5-ms rise and fall times. The spectrum level of the masker was 30-dB SPL. Thresholds for the backward masked tone were determined adaptively for 6 young adults using a rule that converges on the 71% correct point on the psychometric function. The average threshold was 50-dB SPL (SD=12 dB). Thirteen infants were trained to respond to a backward masked 95-dB SPL tone, but not to the masker alone. These infants then completed 30 single-

interval test trials, with 15 no-signal trials and 15 signal trials with the tone fixed at 85-dB SPL and the same backward masker. Infants achieved a $p(C)_{max}$ of about 0.7 ($SD=0.05$) in this task. These results suggest that at this age infants' backward masked threshold is about 35 dB higher than the adults'.

4aSC5. Perceptual cue weighting of voiceless stop consonants as represented by mismatch negativity and P300. Cliff Franklin and Ashley Harkrider (Dept. of Audiol. and Speech Pathol., Univ. of Tennessee, 457 S. Stadium Hall, Knoxville, TN 37996)

Changes in high-frequency content form the basis of differentiating between the perception of voiceless stop consonants, /p/ and /t/. Behavioral studies show that two cues, the formant transition onset frequency and the relative amplitude of the burst component of the stop consonant, contribute to the perception of these consonants, and that the weight of these cues differs for normal-hearing versus hearing-impaired individuals. Our goal in the current study was to obtain a measure of cue weighting electrophysiologically in twelve normal-hearing adults and compare the results with those previously reported in behavioral studies. Two physiological indices of perceptual discrimination of acoustical changes, the attention-independent mismatch negativity (MMN) and the attention-dependent P300, were measured by varying each cue independently and in combination. This resulted in three MMN and three P300 responses per subject in response to (a) changing the formant transition cue with a constant, neutral relative amplitude, (b) changing the relative amplitude cue with a constant, neutral formant transition cue, and (c) changing both cues. Measures from individual and group mean waveforms will be used to determine cue weighting based on sum of square values derived from multivariate analysis of variance. Results will be discussed relative to behavioral perception and attention.

4aSC6. Telephone speech enhancement for hearing-impaired listeners using multi-channel compression. Harikrishna P. Natarajan, Ashok K. Krishnamurthy (Dept. of Elec. Eng., The Ohio State Univ., 2015 Neil Ave., Columbus, OH 43210, krishnamurthy.1@osu.edu), and Lawrence L. Feth (The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210)

Elderly listeners with sensorineural hearing impairment usually have difficulty with telephone communication if they do not use a hearing aid or telephone amplifier. In many cases, these devices may not be readily available or maybe uncomfortable to use. We describe here an alternative strategy to enhance the speech by pre-processing the signal before sending it over the telephone line. The pre-processing is based on a multi-channel compression algorithm that modifies the critical band spectrum of the speech signal to bring it within the dynamic hearing range of the listener, while also compensating for the effects of the telephone line. We describe the algorithm and the results of a simulation study in which the Articulation Index (AI) is used to measure the intelligibility enhancement, and the Glasberg, Moore and Stone (1999) model is used to simulate the effects of the hearing loss.

4aSC7. On learning to recognize spectrally reduced speech: I. The effects of multiple repetitions. Michelle Eng (Dept. of Psych., Univ. of Cincinnati, ML0376, Cincinnati, OH 45221-0376) and Peter Chiu (Univ. of Cincinnati, Cincinnati, OH 45221-0376)

In a previous study [Chiu *et al.*, *J. Acoust. Soc. Am.* **109**, 2501 (2001)], normal listeners learned open-set recognition of words in Harvard IEEE sentences that were spectrally reduced to 4 channels using the CIS processing strategy [Shannon *et al.*, *J. Acoust. Soc. Am.* **104**, 2467–2476 (1998)]. Half of the listeners listened to a male talker during training, the other half a female talker. At different points in the training listeners were tested with blocks of IEEE sentences spoken by the familiar talker, blocks of IEEE sentences by the novel talker, blocks of IEEE sentences of the 2 talkers randomly intermixed, and blocks of HINT sentences. In that study,

listeners could listen to each sentence during early training for an unrestricted number of times before they were given feedback. In the current study, listeners were allowed to listen to each sentence only once during training before feedback was given. Preliminary data obtained from the current study were largely comparable to those obtained in the previous study. The results suggest that massed repetitions and drills, at least in the context of our experimental conditions, have minimal effect on learning to recognize spectrally reduced speech. [Work supported by URC, Univ. of Cincinnati.]

4aSC8. The effects of room acoustics and multiple talkers on speech intelligibility with normal hearing and hearing impaired individuals. Pamela Mishler (Dayton Veterans Affairs Medical Ctr., pamelamishler@wright.edu), Mark Ericson (Wright-Patterson Air Force Base, Dayton, OH), and Shawn Cowell (Dayton Affairs Medical Ctr., Dayton, OH)

Many studies have shown the effect of hearing loss and spatial separation on multi-talker word identification ability in free-field listening conditions. However, little is known about the combination of room reflections and hearing impairment on understanding speech. The purpose of this study was to determine if the effect of room reflections on speech intelligibility was equal among normal hearing and hearing impaired listeners. Two acoustic generation techniques were employed to create the free-field and semi-reverberant listening conditions. Free-field listening conditions were created using virtual acoustic technology. Room reflections were added to the speech samples using dummy-head recordings in a semi-reverberant room. The word identification test used in the experiments was the Modified Rhyme Test (MRT), which afforded a multitude of phonemic pairs when phrases were played simultaneously. Thirty-three listeners in each group participated in each test condition. The hearing impaired listeners were below 60 years of age and exhibited a high frequency, cochlear hearing loss. The normal hearing listeners were age matched with the hearing impaired group. A difference in the type of phonemic identifications between the two techniques and the two groups were found. These phonemic contrasts between will be discussed individually.

4aSC9. The role of reverberation in release from masking due to spatial separation of sources in speech recognition. Gerald Kidd, Jr., Tanya L. Arbogast, Andrew Brughera, Christine R. Mason (Dept. of Commun. Disord. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, gkidd@bu.edu), and William M. Hartmann (Boston Univ., Boston, MA)

Arbogast *et al.* [ARO Mtg. (2002)] found a large release from masking obtained by spatial separation of a target talker and competing speech masker. Both stimuli were sentences from the Coordinate Response Measure corpus [Bolia *et al.*, *J. Acoust. Soc. Am.* (2000)] processed by extracting the envelopes of 15 narrow frequency bands and using the envelopes to modulate carrier tones at the center of each band. By playing nonoverlapping subsets (6–8) of bands from signal and masker they minimized the energetic component while maximizing the informational component of masking. This study extends that work to determine the interaction between reverberation, masker type, and spatial release from masking. Stimuli were processed and presented as above. The target sentence was played at 0-deg azimuth while the masker sentence was played at 0 or 90-deg azimuth. Noise-masker controls were also tested. The listening environment was an IAC booth having dimensions of 12 ft × 13 ft. Acoustic extremes were achieved using Plexiglas™ (highly reflective) or foam (highly absorptive). The results indicated that the amount of masking and the spatial release from masking depend both on the characteristics of the room and masker type. Discussion will center on the acoustic and perceptual factors affecting performance. [Work supported by NIH/NIDCD.]

4aSC10. On learning to recognize spectrally reduced speech: II. Individual differences. Peter Chiu (Dept. of Psych. and Commun. Sci. and Disord., ML0376, Univ. of Cincinnati, Cincinnati, OH 45221-0376), Michelle Eng, Bethany Strange (Univ. of Cincinnati, Cincinnati, OH 45221-0376), Sasha Yampolsky, and Gloria Waters (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215)

When patients with cochlear implants attempt speech recognition, or when listeners with normal hearing try to recognize spectrally reduced speech, performance may suffer because of additional demands on cognitive resources imposed by sensory degradation. Some previous studies have reported a significant correlation between digit span and recognition of spectrally reduced speech, but others have not [Eisenberg *et al.*, *J. Acoust. Soc. Am.* **107**, 2704–2710 (2000)]. In this study, we administered 3 separate measures of working memory capacity (i.e., digit span, alphabet span, and sentence span) and 1 measure of reading ability (American National Adult Reading Test, ANART) to normal-hearing listeners who participated in our previous studies [Chiu *et al.*, *J. Acoust. Soc. Am.* **109**, 2501 (2001) and Fishbeck and Chiu, *J. Acoust. Soc. Am.* **109**, 2504 (2001)]. Preliminary data show that the listener's ability to recognize spectrally reduced speech shows the strongest relationship with ANART scores and the weakest relationship with digit span. Implications of the current findings on theories of speech recognition and working memory will be discussed. [Work supported by URC, Univ. of Cincinnati.]

4aSC11. Role of auditory feedback in speech produced by cochlear implanted adults and children. Sneha V. Bharadwaj, Emily A. Tobey, Peter F. Assmann, and William F. Katz (Callier Ctr. for Commun. Disord., Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235, snehab@utdallas.edu)

A prominent theory of speech production proposes that speech segments are largely controlled by reference to an internal model, with minimal reliance on auditory feedback. This theory also maintains that suprasegmental aspects of speech are directly regulated by auditory feedback. Accordingly, if a talker is briefly deprived of auditory feedback speech segments should not be affected, but suprasegmental properties should show significant change. To test this prediction, comparisons were made between speech samples obtained from cochlear implant users who repeated words under two conditions (1) implant device turned ON, and (2) implant switched OFF immediately before the repetition of each word. To determine whether producing unfamiliar speech requires greater reliance on auditory feedback than producing familiar speech, English and French words were elicited from English-speaking subjects. Subjects were congenitally deaf children ($n=4$) and adventitiously deafened adults ($n=4$). Vowel fundamental frequency and formant frequencies, vowel and syllable durations, and fricative spectral moments were analyzed. Preliminary data only partially confirm the predictions, in that both segmental and suprasegmental aspects of speech were significantly modified in the absence of auditory feedback. Modifications were greater for French compared to English words, suggesting greater reliance on auditory feedback for unfamiliar words. [Work supported by NIDCD.]

4aSC12. Spectral integration of synchronous and asynchronous cues to consonant identification. Joseph W. Hall III, Emily Buss, and John H. Grose (Univ. of North Carolina at Chapel Hill, Dept. of Otolaryngol., CB7070, Chapel Hill, NC 27599-7070)

When speech is presented at poor speech-to-masker ratios, the listener may momentarily receive some spectral regions of the signal at more favorable ratios than other regions. These optimal spectral regions may vary dynamically as a function of time. It is therefore of interest to determine how well listeners utilize cues from different spectral regions that occur asynchronously as opposed to synchronously across frequency. Here, we examined masked identification of consonants under various conditions of 10-Hz square-wave modulation of the speech material. Speech from 100–10 000 Hz was divided into either 2, 4, 8, or 16 contiguous bands. In one set of conditions, bands were amplitude modulated

coherently, while in the other, even and odd bands were modulated 180 degrees out of phase. These conditions, along with further conditions employing only modulated even or odd bands, allowed performance to be compared between synchronous and asynchronous presentation. This approach is similar to a published checkerboard masking study in which the noise rather than the speech was modulated [Howard-Jones and Rosen, *J. Acoust. Soc. Am.* **93**, 2915–2922]. Preliminary data suggest that many listeners are able to utilize asynchronous speech cues efficiently in all of the band number conditions examined. [Work supported by NIDCD (5R01DC000418).]

4aSC13. Relationships among vocabulary size, nonverbal cognition, and spoken word recognition in adults with cochlear implants. Elizabeth A. Collison, Benjamin Munson, and Arlene E. Carney (Dept. of Commun. Disord., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, coll0301@umn.edu)

Recent research has attempted to identify the factors that predict speech perception performance among users of cochlear implants (CIs). Studies have found that approximately 20%–60% of the variance in speech perception scores can be accounted for by factors including duration of deafness, etiology, type of device, and length of implant use, leaving approximately 50% of the variance unaccounted for. The current study examines the extent to which vocabulary size and nonverbal cognitive ability predict CI listeners' spoken word recognition. Fifteen postlingually deafened adults with nucleus or clarion CIs were given standardized assessments of nonverbal cognitive ability and expressive vocabulary size: the Expressive Vocabulary Test, the Test of Nonverbal Intelligence-III, and the Woodcock-Johnson-III Test of Cognitive Ability, Verbal Comprehension subtest. Two spoken word recognition tasks were administered. In the first, listeners identified isophonemic CVC words. In the second, listeners identified gated words varying in lexical frequency and neighborhood density. Analyses will examine the influence of lexical frequency and neighborhood density on the uniqueness point in the gating task, as well as relationships among nonverbal cognitive ability, vocabulary size, and the two spoken word recognition measures. [Work supported by NIH Grant P01 DC00110 and by the Lions 3M Hearing Foundation.]

4aSC14. Speech feature discrimination in deaf children following cochlear implantation. Tonya R. Bergeson, David B. Pisoni, and Karen Iler Kirk (Indiana Univ. School of Medicine, Dept. of Otolaryngol., 699 West Dr. RR044, Indianapolis, IN 46202, tbergeso@iupui.edu)

Speech feature discrimination is a fundamental perceptual skill that is often assumed to underlie word recognition and sentence comprehension performance. To investigate the development of speech feature discrimination in deaf children with cochlear implants, we conducted a retrospective analysis of results from the Minimal Pairs Test (Robbins *et al.*, 1988) selected from patients enrolled in a longitudinal study of speech perception and language development. The MP test uses a 2AFC procedure in which children hear a word and select one of two pictures (bat–pat). All 43 children were prelingually deafened, received a cochlear implant before 6 years of age or between ages 6 and 9, and used either oral or total communication. Children were tested once every 6 months to 1 year for 7 years; not all children were tested at each interval. By 2 years postimplant, the majority of these children achieved near-ceiling levels of discrimination performance for vowel height, vowel place, and consonant manner. Most of the children also achieved plateaus but did not reach ceiling performance for consonant place and voicing. The relationship between speech feature discrimination, spoken word recognition, and sentence comprehension will be discussed. [Work supported by NIH/NIDCD Research Grant No. R01DC00064 and NIH/NIDCD Training Grant No. T32DC00012.]

4aSC15. Understanding speech in single-talker interference: Normal-hearing listeners and cochlear implant users. Peggy B. Nelson and Su-Hyun Jin (Dept. of Commun. Disord., Univ. of Minnesota, 164 Pillsbury Dr. SE, Minneapolis, MN 55455)

Our previous data [Nelson *et al.*, ASA (2001)] indicated that users of cochlear implants experience little release from masking when listening to speech in modulated maskers. Although normal-hearing listeners take advantage of temporal dips in modulated maskers, cochlear implant users do not. The current study investigated listeners with normal hearing and cochlear implants for their understanding of speech in the presence of a single-talker masker. Participants listened to IEEE sentence material spoken clearly at a slow rate (3.5 syllables per second) by either a male (average $F_0=110$ Hz) or a female (average $F_0=210$ Hz) talker. Maskers included continuous speech from other male and female talkers spoken at both slow and fast rates. Results from normal-hearing listeners indicated that when the signal and masker had a similar F_0 and rate, identification was poorer than when signal and masker differed in F_0 and/or rate. The primary factor affecting performance was the F_0 of the masker. It is hypothesized that cochlear implant users will not show the effect of talker/masker F_0 , but will show decreased performance when talker and masker have the same rate. Results from cochlear implant listeners will be reported. [Work supported by NIDCD P01DCD00110.]

4aSC16. Perception of talker similarity by normal-hearing children and hearing-impaired children with cochlear implants. Miranda Cleary (Speech Res. Lab., Dept. of Psych., Indiana Univ., 1101 E. Tenth St., Bloomington, IN 47405)

This project investigated judgments of perceptual similarity, specifically, how similar do two sentences need to be in terms of fundamental and formant frequencies for children to categorize both utterances as spoken by the same talker? Same-different judgments were obtained using an adaptive testing procedure in which the similarity of voice pairs was systematically varied. Participants included five-year-old normal-hearing children and prelingually deafened school-age cochlear implant users with 2+ years of implant experience. Stimuli consisted of natural sentence-length utterances of a fixed syntactic form processed and resynthesized to form a stimulus continuum of differentially similar-sounding voices. Auditory comprehension skills were measured using materials from the same stimulus set. Comprehension was also measured against a competing talker masker. An adaptive testing procedure adjusted the similarity of masker voice to target voice according to each child's performance. The probability of "same talker" judgments is reported for each group of children as a function of pitch difference size. Preliminary results indicate a commonality in the auditory skills necessary for both talker discrimination and auditory comprehension. The relationship between discrimination of similar-sounding talkers and the ability to perceptually separate voices will also be discussed. [Work supported by NIH-NIDCD.]

4aSC17. Auditory streaming in cochlear implant listeners. Monita Chatterjee and John J. Galvin III (Dept. of Auditory Implants and Percept., House Ear Inst., 2100 W. Third St., Los Angeles, CA 90057)

We used the following method to measure auditory streaming in CI listeners: a "test" pattern is composed of two loudness-matched tones A and B presented in the sequence AABAABAAB If A and B are perceptually separable, the subject should hear "AA . . . AA . . . AA . . ." and "B . . . B . . . B . . ." as the two separate streams. The subject hears a "preview" sequence of tones which can have the rhythm of either tone B or tone A in the pattern (e.g., X . . . X . . . X . . . X or XX . . . XX . . . XX . . .), where X is the preview tone. The subject indicates whether or not she/he heard the rhythm of the preview sequence in the test sequence. A number of sets of A, B, and X are set up, and the subject is presented with the different patterns in random order. Results suggest CI listeners are able to perceptually separate tonal sequences into auditory streams. The tonotopic distance (electrode pair separation) between tones is an important factor for streaming. However, listeners are

also able to stream sequences of tones presented to the same electrode pair (tonotopic location) but with different temporal envelopes. [Work supported by NIDCD.]

4aSC18. Cochlear-implant simulations of speech in complex acoustic backgrounds. Michael K. Qin and Andrew J. Oxenham (MIT, 77 Massachusetts Ave., Cambridge, MA 02139-4307)

Cochlear-implant simulations have shown that good speech intelligibility in quiet can be obtained with very poor spectral resolution and no temporal fine-structure information. It is not known how robust speech is to such processing in the presence of complex interfering sounds. This study examined speech reception for sentences in the presence of four types of interference (broadband noise, broadband noise modulated with a speech envelope, a single male talker, and a single female talker) and three processing schemes (24- and 8-channel cochlear-implant simulations and unprocessed). The spectral resolution in the 24-channel condition was designed to be comparable to that of normal hearing, but the original temporal fine structure was replaced by noise in both the 24- and 8-channel processing conditions. Performance with 24-channel processing was substantially poorer than in unprocessed conditions, despite the relatively good frequency resolution. Eight-channel processing produced the poorest performance. Stimulus processing was generally more detrimental to speech reception in the fluctuating interferers than in the unmodulated noise. The detrimental effects of eliminating fine-structure information may be due to the reduction in pitch cues, which listeners may use to help segregate the sources. [Work supported by NIDCD Grant R01DC05216.]

4aSC19. The contribution of auditory temporal processing to the separation of competing speech signals in listeners with normal hearing. Trudy J. Adam and Kathy Pichora-Fuller (Audiol. & Speech Sci., Univ. of BC #226, 5804 Fairview Ave., Vancouver, BC V6T 1Z3, Canada)

The hallmark of auditory function in aging adults is difficulty listening in a background of competing talkers, even when hearing sensitivity in quiet is good. Age-related physiological changes may contribute by introducing small timing errors (jitter) to the neural representation of sound, compromising the fidelity of the signal's fine temporal structure. This may preclude the association of spectral features to form an accurate percept of one complex stimulus, distinct from competing sounds. For simple voiced speech (vowels), the separation of two competing stimuli can be achieved on the basis of their respective harmonic (temporal) structures. Fundamental frequency (F_0) differences in competing stimuli facilitate their segregation. This benefit was hypothesized to rely on the adequate temporal representation of the speech signal(s). Auditory aging was simulated via the desynchronization (~ 0.25 -ms jitter) of the spectral bands of synthesized vowels. The perceptual benefit of F_0 difference for the identification of concurrent vowel pairs was examined for intact and jittered vowels in young adults with normal hearing thresholds. Results suggest a role for reduced signal fidelity in the perceptual difficulties encountered in noisy everyday environments by aging listeners. [Work generously supported by the Michael Smith Foundation for Health Research.]

4aSC20. Identification of musical instruments by normal-hearing subjects listening through a cochlear-implant simulation. Rebecca D. Reich (MIT Media Lab., E15-492 20 Ames St., Cambridge, MA 02139, reich@mit.edu) and Donald Eddington (Cochlear Implant Res. Lab., Boston, MA 02114)

Signal processing in a cochlear implant (CI) is primarily designed to convey speech and environmental sounds, and can cause distortion of musical timbre. Systematic investigation of musical instrument identification through a CI has not yet revealed how timbre is affected by the implant's processing. In this experiment, the bandpass filtering, rectification, and low-pass filtering of an implant are simulated in MATLAB. Syn-

thesized signals representing 12 common instruments, each performing a major scale, are processed by simulations using up to 8 analysis channels. The unprocessed recordings, together with the 8 simulation conditions for 12 instruments, are presented in random order to each of the subjects. The subject's task is to identify the instrument represented by each item. The subjects also subjectively score each item based on similarity and pleasantness. We anticipate performance using the simulation will be worse than the unprocessed condition because of the limited information delivered by the envelopes of the analysis channels. These results will be analyzed as a confusion matrix and provide a basis for contrasting the information used by subjects listening to the unprocessed and processed materials. Understanding these differences should aid in the development of new processing strategies to better represent music for cochlear implant users.

4aSC21. Typicality ratings of male and female voices. Brian Spisak, John Mullenix, Kelly Moro, Jessica Will (Dept. of Psych., Univ. of Pittsburgh, Johnstown, Johnstown, PA 15904), and Lynn Farnsworth (Wayne State Univ., Detroit, MI 48202)

Researchers have suggested that human voices are represented in memory in terms of prototypes [e.g., Kreiman and Papcun (1991); Papcun *et al.* (1989)]. Others have suggested that speech utterances are stored in memory via detailed exemplar-based representations [e.g., Lachs *et al.* (2000)]. The goal of the present study was to provide the first step toward assessing the viability of a prototype view of voice. Ten hVd utterances were recorded from each of 20 male and 20 female speakers. The utterances were blocked by speaker gender and presented to male and female listeners who rated each stimulus on a 1–7 typicality scale from “least typical voice” to “most typical voice.” There were significant effects of the type of vowel and speaker voice on the ratings, as well as interactions of vowel type with gender of subject and speaker voice. The results are discussed in terms of the strength of evidence for a graded category structure of voice categories that would be consistent with a prototype perspective of long-term memory representations of voice.

4aSC22. Study of effect of low-pass filtering for channels amplitude estimation on speech intelligibility by acoustic simulation of SPEAK strategy. Erdenebat Dashtseren and Shigeyoshi Kitazawa (Grad. School of Sci. and Eng., Johoku 3-5-1 Hamamatsu, Shizuoka 432-8011, Japan, bat@cs.inf.shizuoka.ac.jp)

The conventional way of amplitude envelope estimation for stimulation pulse train generation for selected channels within multi-channel cochlear implant speech processors is low-pass filtering of half or full wave rectified (HWR or FWR) frequency bands of the corresponding channels. To study the necessity of low-pass filtering (LPF) for amplitude envelope estimation, acoustic simulation of the SPEAK speech strategy was implemented. Stimulation pulse patterns for selected channels are estimated without LPF and compared to those estimated by the conventional way. Acoustic stimuli are regenerated using an overlap adding technique of filtered pulse trains. Speech intelligibility tests on CV and VCV of Japanese within normal-hearing subjects were conducted to measure the effect of LPF. A one-way ANOVA indicates that the LPF with a cutoff frequency 160 Hz for amplitude envelope estimation of selected channels negatively affected the speech intelligibility.

4aSC23. Further investigations on binaural intelligibility of reverberant speech. Brad Libbey and Peter Rogers (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, gt1556a@prism.gatech.edu)

There is a binaural intelligibility advantage for many subjects when listening to reverberant speech. Two processes offer explanations as to why the binaural advantage exists. First, directional information may identify echoes and remove a portion of the reverberation in a manner similar

to the precedence effect. A more likely possibility is that the reverberation behaves as additive noise with a specific time and frequency dependence. In this case, subjects may utilize the coherence of the left and right channels to limit the effects of the incoherent reverberation, similar to binaural masking release. To test the latter hypothesis, subjects listen to diotic (L+R), binaural (L&R), and diotic (L or R) reverberant words. The diotic treatment (L+R) adds two recordings measured at the location of the listener's ears. Since the listener faces the source, the direct signals are coherent while some cancellation is expected for the reverberation. The presentation will compare diotic (L+R) intelligibility to the binaural (L&R) condition as well as the diotic (L or R) control condition where the left or right signal plays to both ears.

4aSC24. The perception of voice pitch by naïve and experienced listeners. Donald Fucci (School of Hearing, Speech and Lang. Sci., Ohio Univ., Grover Ctr. W222, Athens, OH 45701, fucci@ohio.edu), Marianne Schueller, and Zinny Bond (Ohio Univ., Athens, OH 45701)

Perceptual judgments of a single speaker's voice pitch will be characterized by any given listener along a subjective continuum. Individual listener attributes have been suggested as an explanation for this diversity in categorizing voice pitch. Professionals involved in the evaluation of voice, specifically voice pitch, are expected to possess a level of perceptual skill that reliably quantifies pitch regardless of personal attributes. Two groups of individuals, one naïve and one experienced, were asked to perceptually match voice pitch during two pitch matching tasks. The first task involved matching an audio taped voice pitch produced on the vowel /a/ to a musical note on a keyboard. The second task involved listening to an audio-taped voice producing words, then matching the voice pitch of the word to a musical note on a keyboard. Accuracy of perceptual judgments was measured upon stimuli produced and assigned musical keyboard note by master's level opera music majors. Results indicate that a difference exists between groups in only one of the four variables measured. Individual differences varied, but appeared unrelated to previous musical experience.

4aSC25. Peripheral and central locus of a nonspeech phonetic context effect. Sarah C. Sullivan and Andrew J. Lotto (Washington State Univ., Pullman, WA 99164-4820, alotto@wsu.edu)

Previous work has demonstrated that nonspeech sounds with the appropriate spectral characteristics can affect the identification of speech sounds [Lotto and Kluender, *Percept. Psychophys.* **60**, 602–619 (1998)]. It has been proposed that these spectral context effects are due to interactions in the peripheral auditory system. For example, they could be the result of masking at the auditory nerve or of auditory enhancement effects that have been demonstrated to be monaural [Summerfield and Assmann, *Percept. Psychophys.* **45**, 529–536 (1989)]. To examine the locus of the context effect, synthesized syllables varying from /da/ to /ga/ were preceded by single-formant stimuli that mimicked the third formant of the syllables /a/ and /ar/. The nonspeech stimulus was presented either to the same or opposite ear as the target speech stimulus. Subjects speech identifications were shifted as a function of context in predicted directions for both presentation conditions. However, the size of the shift was smaller when the context was in the ear contralateral to the target syllable. These results agree well with similar results for speech contexts. The data suggest that the context effects occur at multiple levels of the auditory system and are not simply examples of masking or auditory enhancement.

4aSC26. Reverberation reduction by sub-band unmasking. Brad Libbey and Peter Rogers (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, gt1556a@prism.gatech.edu)

Reverberation reduces the intelligibility of speech recorded in rooms. Signal processing algorithms attempt to reduce this reverberation such that the speech not only sounds less reverberant, but is more intelligible. His-

torically, these techniques are based on calculations of the room impulse response, on arrays of two or more microphones, noise cancellation, or cepstral techniques. The new technique assumes that the intelligibility loss is due to overlap masking of reverberant tails of speech. The algorithm looks at sub-bands and identifies those that mask adjacent bands. When this occurs in conjunction with a decaying envelope the decay rate is increased. Thus, the masked band has a greater chance of being heard. Intelligibility results for the processed speech will be presented.

4aSC27. Effect of preceding speech on nonspeech sound perception.

Joseph D. Stephens and Lori L. Holt (Psych. Dept., Carnegie Mellon Univ., and Ctr. for the Neural Basis of Cognition, 5000 Forbes Ave., Pittsburgh, PA 15213)

Data from Japanese quail suggest that the effect of preceding liquids (/l/ or /r/) on response to subsequent stops (/g/ or /d/) arises from general auditory processes sensitive to the spectral structure of sound [A. J. Lotto, K. R. Kluender, and L. L. Holt, *J. Acoust. Soc. Am.* **102**, 1134–1140 (1997)]. If spectral content is key, appropriate nonspeech sounds should influence perception of speech sounds and vice versa. The former effect has been demonstrated [A. J. Lotto and K. R. Kluender, *Percept. Psychophys.* **60**, 602–619 (1998)]. The current experiment investigated the influence of speech on the perception of nonspeech sounds. Nonspeech stimuli were 80-ms chirps modeled after the F_2 and F_3 transitions in /ga/ and /da/. F_3 onset was increased in equal steps from 1800 Hz (/ga/ analog) to 2700 Hz (/da/ analog) to create a ten-member series. During AX discrimination trials, listeners heard chirps that were three steps apart on the series. Each chirp was preceded by a synthesized /al/ or /ar/. Results showed context effects predicted from differences in spectral content between the syllables and chirps. These results are consistent with the hypothesis that spectral contrast influences context effects in speech perception. [Work supported by ONR, NOHR, and CNBC.]

4aSC28. Recognition masking for transitions in initial and final position.

Jeremy R. Gaston and Richard E. Pastore (Dept. of Psych., Binghamton Univ., P.O. Box 6000, Binghamton, NY 13901)

There is a common implicit assumption that the input to higher or specialized processes is accurately represented by the physical properties of the stimulus, with, at most, only detection masking and some basic initial filtering. Since playing an auditory stimulus backwards does not alter the spectral composition of the stimuli, this initial assumption has led to the conclusion that any differences in perception must be attributed to higher level or specialized processes. Thus, for example, significant differences in the perception of temporally reversed CV syllables are assumed to reflect only specialized processes (i.e., speech module). The current study challenges this basic implicit assumption. Backward “recognition” masking is a basic, but poorly understood perceptual phenomenon in which an added following stimulus interferes with discrimination of simple preceding tones. Recent work from our laboratory indicates similar interference in the recognition of easily discriminated chords. The current study investigates the role of backward recognition masking in determining ability to recognize the direction of frequency glides similar to formant transitions of stop consonants.

4aSC29. Acoustical considerations for an automotive hands-free mobile terminal.

Richard J. Ruhala and Vasu Iyengar (Agere Systems, 1247 S. Cedar Crest Blvd., Allentown, PA 18103, ruhala@agere.com)

The harsh acoustical environment of an automotive interior yields significant difficulty in controlling acoustic echo and noise during “hands-free” wireless phone calls. In addition to having multiple closely coupled reflection paths, the user (talker) is often positioned between the loudspeaker and microphone path. This can cause frequent changes in the acoustic path which necessitates fast and accurate adaptation of the acoustic echo canceller and/or non-linear controller. Further, the frequency re-

sponse between the talker and the microphone is altered due to cabin resonances and modes; it is also strongly affected by their positions. Additional acoustical considerations for automotive hands-free implementations are the high levels and non-stationary statistics of background noise, high amplification, and possible resultant distortion of the received audio signal, and non-linear network echo. These impairments yield significant challenges in controlling the acoustic echo and noise in full-duplex, partial-duplex, and even half-duplex systems. In this work, impulse response, frequency response, and SNR measurements are presented and analyzed to demonstrate the critical differences between a traditional desktop speakerphone and an automotive hands-free mobile terminal.

4aSC30. Acoustic and perceptual effects of overall F_0 range in a lexical pitch accent distinction.

Travis Wade (Dept. of Linguist., Univ. of Kansas, 1541 Lilac Ln., 427 Blake Hall, Lawrence, KS 66044-3177)

A speaker’s overall fundamental frequency range is generally considered a variable, nonlinguistic element of intonation. This study examined the precision with which overall F_0 is predictable based on previous intonational context and the extent to which it may be perceptually significant. Speakers of Tokyo Japanese produced pairs of sentences differing lexically only in the presence or absence of a single pitch accent as responses to visual and prerecorded speech cues presented in an interactive manner. F_0 placement of high tones (previously observed to be relatively variable in pitch contours) was found to be consistent across speakers and uniformly dependent on the intonation of the different sentences used as cues. In a subsequent perception experiment, continuous manipulation of these same sentences between typical accented and typical non-accent-containing versions were presented to Japanese listeners for lexical identification. Results showed that listeners’ perception was not significantly altered in compensation for artificial manipulation of preceding intonation. Implications are discussed within an autosegmental analysis of tone. The current results are consistent with the notion that pitch range (i.e., specific vertical locations of tonal peaks) does not simply vary gradually across speakers and situations but constitutes a predictable part of the phonetic specification of tones.

4aSC31. Comparison of approaches to estimate the speech modulation transfer function.

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Using speech as a probe stimulus to compute the Speech Transmission Index (STI) has been of great interest to speech researchers. One technique is based on first computing the speech modulation transfer function (SMTF). Approaches used to obtain the SMTF include those developed by Steeneken and Houtgast [H. Steeneken and T. Houtgast, *Proc. 11th ICA*, Paris **7**, 85–88 (1983)] and Drullman *et al.* [R. Drullman, J. M. Festen, and R. Plomp, *J. Acoust. Soc. Am.* **95**, 2670–2680 (1994)]. This paper compares these two approaches and a new one to the theoretically obtained modulation transfer function (MTF) for reverberant and noisy environments. The new method computes the magnitude of the cross-power spectrum rather than the real part used by Drullman. As previously reported, Houtgast’s method exhibits artifacts at high modulation frequencies. Drullman’s approach eliminates artifacts in the reverberant environment but does not predict the theoretical MTF for the noisy environment. The new method outperforms the other two approaches in matching the theoretically derived MTF across both environments. This paper also examines the SMTF of amplitude-compressed speech for these three methods. [Work supported by NIDCD.]

4aSC32. The effect of variation in naturalness on phonetic perceptual identification. Robert E. Remez, Cynthia Y. Yang (Dept. of Psych., Barnard College, 3009 Broadway, New York, NY 10027-6598, remez@columbia.edu), Rebecca L. Piorkowski (Johns Hopkins Univ., Baltimore, MD 21218), Stephanie Wissig, Abigail Batchelder, and Heddy Nam (Barnard College, New York, NY 10027-6598)

The relation between apparent naturalness and phonetic identification was assessed in six perceptual tests. A seven-step place-of-articulation series spanning [da] to [ga] was created with speech synthesis approximating the spectra of natural samples. The sensitivity of perceivers to this realization of a place contrast was assessed by estimating the cumulative d' across the series in identification tests. Four variants of this series differing in apparent naturalness were produced by altering the synthesis source function while preserving the center frequency and bandwidth of the formants, and by replicating the gross spectrotemporal patterns with time-varying sinusoids. In addition to calibrating perceivers' sensitivity to the place contrast over variation in naturalness, we conducted a naturalness tournament composed of items drawn from the five test series. A correlation of the findings of the naturalness tournament with the measures of phonetic sensitivity offers an index of the effect of variation in naturalness on phonetic perception. This study can resolve the dispute between the classic premise that intelligibility and naturalness are orthogonal attributes of speech perception, and the more recent premise entailed by episodically based accounts of perceptual categorization, that novel instances are identified by virtue of auditory similarity to prior exemplars. [Research supported by NIDCD.]

4aSC33. Effects of bandpass noise and telephonic filtering on the perception of consonants. Thomas Purnell and Laura Kopplin (Dept. of Linguist., Univ. of Wisconsin, Madison, 1168 Van Hise, 1220 Linden Dr., Madison, WI 53706, tpurnell@facstaff.wisc.edu)

Consonant identification studies report varying results of the usefulness of information above 4 kHz. A recent study found listeners misidentifying female-produced consonants characterized by a center of gravity above 3.5 kHz [P. G. Stelmachowicz *et al.*, *J. Acoust. Soc. Am.* **110**, 2183–2190 (2000)], suggesting that the perception of alveolar stops and fricatives as spoken by females are misperceived in common environments which filter out high-frequency spectral information. In the current study, consonant identification of female tokens consisting of consonants varying by place or manner ([p t k], [f T s S], [t s S n l r]) was tested in a forced choice experiment. Tokens were low-pass filtered at 3.5 kHz simulating telephonic filtering. Additionally, the locus of spectral information below 4 kHz was assessed by comparing responses to tokens masked by nonmodulating noise, where the signal-to-noise ratio was at least 10 dB SPL. Noise masking occurred between 0 and 0.4, 0.4 and 2, and 2 and 3.5 kHz. Confusion matrices indicated that the perception of unmasked speech is not hindered by low-pass filtering at 3.5 kHz, while perception of masked tokens was worse with masking between 2 and 3.5 kHz.

4aSC34. Consonant identification in noise: Effects of temporal asynchrony, noise fluctuations, and temporal envelope expansion. Frederic Apoux (LPE, UMR CNRS 8581, Inst. de Psychologie, Univ. Paris V, 71 Av E. Vaillant, 92774 Boulogne-Billancourt Cedex, France and INTRASON France S.A.), Stephane Garnier (Groupement d'audioprothésistes ENTENDRE-GIPA 2, 78760 Jouars-Pontchartrain, France), and Christian Lorenzi (Inst. de Psychologie, Univ. Paris V, 92774 Boulogne-Billancourt Cedex, France)

The effects of two temporal envelope expansion schemes on consonant identification performance were investigated with normal-hearing listeners. These effects were examined in the presence or absence of short-term energy fluctuations in the masker. The effects of adding a 500-ms forward fringe to the background noise was also investigated. Speech stimuli were 16 consonant-*/a/* syllables. The expanded stimuli were obtained by raising their original envelope to the power k . In the first scheme, k was fixed at 2. In the second scheme, k varied from 4 to 0.3 as a function of the

envelope power; the low-level parts of the envelope were therefore expanded whereas the high-level parts were compressed. As reported in many studies, identification was substantially improved by the presence of energy fluctuations in the masker. Adding the forward fringe generally degraded identification in fluctuating noise, but could improve identification in the stationary noise. Overall, expansion led to (i) poorer consonant identification when k was a constant and (ii) better consonant identification when k was varied. These effects were larger in stationary noise. This study suggests that consonant identification can be improved by a dual expansion/compression scheme. Some results obtained with hearing-impaired listeners will be presented.

4aSC35. Recognition of word and nonsense syllables in noise. José R. Benkí (Dept. of Linguist., Univ. of Michigan, Ann Arbor, MI 48109-1285)

In the recognition of human speech, listeners use sensory information from the speech signal as well as signal-independent information, such that acoustic-phonetic salience, lexical status, frequency of usage, and neighborhood density interact in speech recognition. This paper presents detailed results from a replication of a study of context effects reported by Boothroyd and Nittrouer [*J. Acoust. Soc. Am.* **84**, 101–108 (1988)]. 240 phonetically matched word and nonsense CVC syllables at different SNRs were presented to 32 listeners for identification. The results are consistent with the original study, with greater accuracy for words than nonsense items as quantified by a j -factor analysis according to lexical status. Online response collection enabled detailed analyses not reported in the original study. Values for the j -factor according to usage frequency and phonetic confusability are presented. Confusion matrices of the phonemes are presented, and are largely consistent with previous studies for the initial consonants and vowels. Accuracy in the final consonants of nonsense syllables is substantially lower than the corresponding phonemes of words or than the initial consonants of nonsense syllables. Final nasals were particularly difficult to identify.

4aSC36. Signal detection analysis of response distributions for intensity and speech judgments. Richard E. Pastore (SUNY at Binghamton, Binghamton, NY 13902-6000) and Neil A. Macmillan (Dept. of Psych., Brooklyn College (CUNY), Brooklyn, NY 11210-2889)

Schouten and van Hensen [*J. Acoust. Soc. Am.* **104**, 2980–2990 (1998)] reported rating response distributions for intensity and speech continua. These empirical distributions were claimed to reflect each subject's internal decision axes, allowing direct evaluation of the distribution assumptions underlying the common Gaussian equal-variance model signal detection theory (SDT). The observed distributions were deemed roughly Gaussian. However, the original analyses implicitly assume that the rating scale is a direct, interval reflection of internal likelihood values. We used the data instead to construct ROC curves, an analysis that assumes an ordinal relationship between the rating responses and the decision axes. Our ROC analyses show that the representations of the intensity stimuli reasonably approximate a Gaussian model. In contrast, the distributions underlying the speech stimuli are clearly not Gaussian, but imply discrete representations. We discuss appropriate decision models for judgments of intensity and speech continua.

4aSC37. Speech perception with tactile support in adverse listening conditions. Rob Drullman and Adelbert W. Bronkhorst (TNO Human Factors, P.O. Box 23, 3769 ZG Soesterberg, The Netherlands)

Since long, different methods of vibrotactile stimulation have been used as an aid for speech perception by some people with severe hearing impairment. The fact that experiments have shown (limited) benefits proves that tactile information can indeed give some support. In our research program on multimodal interfaces, we wondered if normal hearing listeners could benefit from tactile information when speech was presented

in adverse listening conditions. Therefore, we set up a pilot experiment with a male speaker against a background of one, two, four or eight competing male speakers or speech noise. Sound was presented diotically to the subjects and the speech-reception threshold (SRT) for short sentences was measured. The temporal envelope (0–30 Hz) of the speech signal was computed in real time and led to the tactile transducer (MiniVib), which was fixed to the index finger. First results show a significant drop in SRT of about 3 dB when using tactile stimulation in the condition of one competing speaker. In the other conditions no significant effects were found, but there is a trend of a decrease of the SRT when tactile information is given. We will discuss the results of further experiments.

4aSC38. Audiovisual integration in the absence of a McGurk effect.

Lawrence Brancazio (Dept. of Psych., Southern Connecticut State Univ., 501 Crescent St., New Haven, CT 06515 and Haskins Labs., New Haven, CT 06511), Joanne L. Miller, and Michèle Mondini (Northeastern Univ., Boston, MA 02115)

The McGurk effect, a change in perceived place of articulation due to an incongruent visual stimulus (e.g., auditory /pi/ with visual /ti/ perceived as /ti/), demonstrates the contribution of vision to speech perception. Interestingly, in a given experiment the McGurk effect typically does not occur on every trial. We investigated whether non-McGurk trials result from a failure to perceptually integrate auditory and visual information by simultaneously manipulating visual place of articulation and visual speaking rate. Previous work [Green and Miller, *Percept. Psychophys.* **38**, 269–276 (1985)] has shown that the boundary along an auditory /bi-/pi/ voice-onset-time (VOT) continuum occurs at a longer VOT when the auditory stimulus is paired with a slow rather than a fast visual /pi/. We paired stimuli from an auditory /bi-/pi/ continuum with fast and slow versions of a visual /ti/, and subjects identified each item as /b/, /p/, /d/, or /t/. We found a rate effect on McGurk trials, with the /d-/t/ boundary occurring at a longer VOT when the visual stimulus was slow rather than fast. Importantly, we found a comparable rate effect for the /b-/p/ boundary on non-McGurk trials. This indicates that audiovisual integration occurs even in the absence of a McGurk effect. [Work supported by NIH/NIDCD.]

4aSC39. Elderly perception of speech from a computer.

Alan Black, Maxine Eskenazi, and Reid Simmons (Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213)

An aging population still needs to access information, such as bus schedules. It is evident that they will be doing so using computers and especially interfaces using speech input and output. This is a preliminary study to the use of synthetic speech for the elderly. In it twenty persons between the ages of 60 and 80 were asked to listen to speech emitted by a robot (CMU's VIKIA) and to write down what they heard. All of the speech was natural prerecorded speech (not synthetic) read by one female speaker. There were four listening conditions: (a) only speech emitted, (b) robot moves before emitting speech, (c) face has lip movement during speech, (d) both (b) and (c). There were very few errors for conditions (b), (c), and (d), but errors existed for condition (a). The presentation will discuss experimental conditions, show actual figures and try to draw conclusions for speech communication between computers and the elderly.

4aSC40. Virtual glides and duplex perception.

Robert Allen Fox, Lawrence L. Feth, Ewa Jacewicz (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu), and Nandini Iyer (Univ. of Illinois, Champaign, IL 61820)

Duplex perception occurs when a tone glide is presented to one ear and an ambiguous synthetic CV is presented to the other ear. Listeners simultaneously perceive a syllable (e.g., /da/ or /ga/) and a tone chirp. The

present study examines whether duplex percepts can be generated when the tone glide presented is actually a virtual glide in which the frequency transition is evoked by modifying the amplitudes of two tones whose frequencies do not change [see Jacewicz *et al.*, *J. Acoust. Soc. Am.* **110**, 2657 (2002)]. A base synthetic CV token 250 ms in duration (with initial *F1* and *F2* transitions appropriate to both /d/ and /g/) was combined with a set of *F3* transitions (representing either a synthetic formant transition, a true tone glissando, or a virtual glide transition) which produced a set of /da-/ga/ continua. These *F3* transitions were either combined with the base and presented to both ears (diotic presentation) or presented simultaneously to different ears (dichotic presentation). Listeners heard the stimulus sets and were required to identify each token as /da/ or /ga/. Preliminary results indicate that similar identification functions for all versions of *F3* are generated in the monaural condition, but that these functions are shallower in the dichotic condition.

4aSC41. Relative roles of consonants and vowels in perceiving phonetic versus talker cues in speech.

Gina Cardillo and Michael J. Owren (Dept. of Psych., Cornell Univ., Ithaca, NY 14853)

Perceptual experiments tested whether consonants and vowels differentially contribute to phonetic versus indexical cueing in speech. In 2 experiments, 62 total participants each heard 128 American-English word pairs recorded by 8 male and 8 female talkers. Half the pairs were synonyms, while half were nonsynonyms. Further, half the pairs were words from the same talker, and half from different, same-sex talkers. The first word heard was unaltered, while the second was edited by setting either all vowels ("Consonants-Only") or all consonants ("Vowels-Only") to silence. Each participant responded to half Consonants-Only and half Vowels-Only trials, always hearing the unaltered word once and the edited word twice. In experiment 1, participants judged whether the two words had the same or different meanings. Participants in experiment 2 indicated whether the word pairs were from the same or different talkers. Performance was measured as latencies and *d* values, and indicated significantly greater sensitivity to phonetic content when consonants rather than vowels were heard, but the converse when talker identity was judged. These outcomes suggest important functional differences in the roles played by consonants and vowels in normative speech.

4aSC42. Models of vowel perception.

Michelle R. Molis^{a)} (Army Audiol. & Speech Ctr., Walter Reed Army Medical Ctr., Washington, DC 20307-5001)

The debate continues regarding the efficacy of formant-based versus whole spectrum models of vowel perception. Categorization data were obtained for a set of synthetic steady-state vowels and were used to evaluate both types of models. The models tested included various combinations of formant frequencies and amplitudes, principal components derived from critical-band spectra, and perceptually scaled LPC cepstral coefficients. The stimuli were 54 five-formant synthesized vowels that varied orthogonally in *F2* (1081–2120 Hz) and *F3* (1268–2783 Hz) frequency in equal 0.8 Bark steps. Fundamental frequency contour, *F1* (455 Hz), *F4* (3250 Hz), *F5* (3700 Hz), and duration (225 ms) were held constant across all stimuli. Twelve speakers of American English (Central Texas dialect) categorized the stimuli as the vowels /I/, /U/, or /e/. Results indicate that formant frequencies provided the best account of the data only when nonlinear terms were also included in the analysis. The critical-band principal components also performed reasonably well. While the principle of parsimony would suggest that formant frequencies offer the most appropriate description of vowels, the relative success of a richer,

more flexible and more neurophysiologically implementable model still must be taken into consideration. [Work supported by NIDCD.]
a) Formerly at Univ. of Texas at Austin.

4aSC43. Patterns in the perception of VC(C)V strings. Terrance M. Nearey (Dept. of Linguist., Univ. of Alberta, Edmonton, AB T6G 2E7, Canada, t.nearey@ualberta.ca) and Roel Smits (Max Planck Inst. for Psycholinguist.)

A set of 144 /aC(C)a/ stimuli were constructed (inspired by, e.g., B. Repp, *Percept. Psychophys.* **24**, 471–485 (1978)). Each consisted of: a vocoid (135 ms steady state followed by 50 ms VC closing transitions); a silent portion; and another vocoid (50 ms CV opening transition plus 250 ms steady state). Four different silent periods [80, 120, 190, and 300 ms] were crossed with six closing transition patterns (ranging from b-like to d-like closures) and with six analogous opening transitions. Thirteen listeners classified each stimulus ten times. Polytomous logistic regression showed that singletons, heterorganic clusters, and geminates all have distinct duration weights. Contrary to expectations, results for place were remarkably simple. Weights for closing transitions show labial closures (/b, bd, bb/) to be widely separated from the coronal closures (/d, db, dd/), but are otherwise homogeneous. Weights for opening cues show a similar dichotomy of labial releases /b, db, bb/ versus coronal releases (/d, bd, dd/); here, however, the singleton /b/ differs slightly from the other labials. A related experiment in Dutch will be discussed and results from both experiments will be compared against proposals from the literature and our own models. [Work supported by SSHRC.]

4aSC44. The effect of monauralized binaural displays on speech intelligibility. Robert S. Bolia (Air Force Res. Lab. (AFRL/HECP), 2255 H St., Wright-Patterson AFB, OH 45433-7022, robert.bolia@wpafb.af.mil)

It has often been demonstrated in laboratory settings that the spatial separation of multiple simultaneous talkers leads to enhanced intelligibility of a target message as well as an increase in the ease with which nontarget messages can be successfully ignored. This finding can be exploited in binaural intercom systems to improve communications' effectiveness. In some situations, operators use monaural displays at least part of the time, for example when operators in command and control nodes remove one earcup to talk to other nearby operators, in an effort to avoid squandering channels. The present investigation was designed to determine whether such a temporary monauralization of a binaural intercom would result in reduced speech intelligibility relative to a diotic control condition. Participants listened to 2 or 3 simultaneous presentations of phrases from the coordinate response measure (Bolia, Nelson, Ericson, and Simpson, 2000) and responded according to the instructions spoken by the target talker. The dependent measure was the percentage of correct responses. Independent variables included listening condition (monaural left, monaural right, binaural) and the locations of the target and distractor talkers. Results indicate that the performance in the monaural conditions were worse than in the binaural conditions but better than the diotic control.

4aSC45. Cerebral responses to across- and within-category change of vowel durations measured by near-infrared spectroscopy. Yasuyo Minagawa-Kawai (Independent Administrative Inst., Natl. Inst. of Japanese Lang.), Koichi Mori, Izumi Furuya, Ryoko Hayashi, and Yutaka Sato (Res. Inst., Natl. Rehabilitation Ctr. for the Disabled)

The present study examined cerebral responses to phoneme categories, using near-infrared spectroscopy (NIRS) by measuring the concentration and oxygenation of hemoglobin accompanying local brain activities. Targeted phonemes used here are Japanese long and short vowel categories

realized only by durational differences. Results of NIRS and behavioral test revealed NIRS could capture phoneme-specific information. The left side of the auditory area showed large hemodynamic changes only for contrasting stimuli between which phonemic boundary was estimated (across-category condition), but not for stimuli differing by an equal duration but belonging to the same phoneme category (within-category condition). Left dominance in phoneme processing was also confirmed for the across-category stimuli. These findings indicate that the Japanese vowel contrast based only on duration is dealt with in the same language-dominant hemisphere as the other phonemic categories as studied with MEG and PET, and that the cortical activities related to its processing can be detected with NIRS. [Work supported by Japan Society for Promotion of Science (No. 8484) and a grant from Ministry of Health and Welfare of Japan.]

4aSC46. Importance of onset properties on spectral contrast for speech and other complex spectra. Jeffry A. Coady and Keith R. Kluender (Dept. of Psych., Univ. of Wisconsin, Madison, Madison, WI 53706, jacoody@facstaff.wisc.edu)

Previous studies using speech and nonspeech analogs have shown that auditory mechanisms that enhance spectral contrast serve to compensate for assimilation of spectral composition caused by coarticulated production. In the present studies, series of CV syllables varying acoustically in F_2 -onset frequency and perceptually from /ba/ to /da/ were identified either following [e] (higher F_2) and [o] (lower F_2) or following spectral complements of these vowels. Spectral complements had troughs replacing peaks corresponding to formants in the vowels. Both preceding vowels and spectral complements altered perception of the following stop, with complementary spectra providing results complementary to those for vowel spectra. However, effects for preceding spectral complements differed from those for preceding vowels by being dependent upon the precursor duration or inter-stimulus interval (ISI). Additional experiments using short spectral samples revealed that the single most important predictor of effects of vowel complements was the interval between the onset of the preceding sound and the onset of the following CV. This finding appears to implicate spectral characteristics of onsets (in addition to or instead of simple adaptation or masking) as being important for spectral contrast. The significance of these findings for the perception of connected speech will be described. [Work supported by NIDCD.]

4aSC47. Center-of-gravity effects in the perception of high front vowels. Ewa Jacewicz and Lawrence L. Feth (Dept. of Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, jacewicz.1@osu.edu)

When two formant peaks are close in frequency, changing their amplitude ratio can shift the perceived vowel quality. This center-of-gravity effect (COG) was studied particularly in back vowels whose F_1 and F_2 are close in frequency. Chistovich and Lublinskaja (1979) show that the effect occurs when the frequency separation between the formants does not exceed 3.5 bark. The COG and critical distance effects were manifested when a two-formant reference signal was matched by a single-formant target of variable frequency. This study investigates whether the COG effect extends to closely spaced higher formants as in English /i/ and /I/. In /i/, the frequency separation between F_2 , F_3 , and F_4 does not exceed 3.5 bark, suggesting the existence of one COG which may affect all three closely spaced formants ($F_2 = 2030$, $F_3 = 2970$, $F_4 = 3400$ Hz). In /I/, each of the F_2 - F_3 and F_3 - F_4 separations is less than 3.5 bark but the F_2 - F_4 separation exceeds the critical distance, indicating two COGs ($F_2 = 1780$, $F_3 = 2578$, $F_4 = 3400$ Hz). We examine the COG effects using matching of four-formant reference signals, in which we change the

amplitude ratios, by two-formant targets with variable frequency of F_2 . The double-staircase adaptive procedure is used. [Work supported by an INRS award from NIH to R. Fox.]

4aSC48. Speaking rate adjustment across changes in talker. Rochelle S. Newman (Dept. of Speech & Hearing Sci., Univ. of Maryland, 0100 Lefrak Hall, College Park, MD 20742) and James R. Sawusch (Univ. at Buffalo, Buffalo, NY 14260)

Individuals vary their speaking rate, and listeners use the speaking rate of precursor sentences to adjust for these changes [Kidd (1989)]. Recent work has suggested that speaking rate adjustment may not always be limited to speech from a single talker (Sawusch and Newman, 2000). When a talker change occurs in the midst of a vowel, the durations of both portions of the vowel influence perception of the rate-based contrast. In the present experiments, we examined the effect of talker change on rate normalization for precursor phrases. A male talker produced the sentence, I heard him say the word—at one of three different speaking rates. A female talker then produced a nonword containing a duration-based contrast. We examined whether the male talker's speaking rate would influence perception of the female talker's speech. The results were somewhat surprising. The speaking rate of the first talker did influence perception of the contrast in the second talker. However, the effect was a U-shaped function of speaking rate, rather than the linear function typically demonstrated in the single-voice condition. Several follow-up studies replicated this basic pattern. Implications of this finding for studies of rate normalization will be discussed.

4aSC49. The effect of familiarity with a voice on continuous shadowing: Effects of explicit knowledge, but not of implicit familiarity. Rochelle S. Newman (Dept. of Hearing & Speech Sci. & Ctr. for Neurosci. & Cognit. Sci., Univ. of Maryland, College Park, MD 20742)

Listeners were asked to shadow a target voice while a second voice spoke fluently in the background. Listeners differed in their familiarity with the target voice: members of the first group of listeners were familiar with the voice and were told explicitly whose voice they would be hearing; members of a second group were familiar with the target voice but were not warned whose voice it was, and members of a third group were entirely unfamiliar with the target voice. Familiarity was from real-world experience: the voice was that of one of the participant's professors. Explicit knowledge of talker identity appeared to have the larger effect on listener performance: Participants in the first group made significantly fewer shadowing errors than those in the second group. In contrast, simple familiarity with a voice had no such influence. Familiarity did influence the types of errors listeners made, however: those participants who were familiar with the target voice prior to the test session made fewer incorrect responses than did those who had not heard the talker previously, although their total number of errors (including misses, as well as incorrect responses) did not differ.

4aSC50. Optimization of a three-dimensional Simplex procedure for speech in noise. Bas A. M. Franck, Jan Koopman, and Wouter A. Dreschler (Academic Medical Ctr., Dept. of Clinical and Exp. Audiol., D2-223, P.O. Box 1100 DD, Amsterdam, The Netherlands)

In this study we investigated means to enhance the reliability of a three-dimensional Simplex optimization procedure. The algorithms used to span the three dimensions are noise reduction, spectral tilting, and spectral enhancement, respectively. We considered the influence of an adaptive

step size, different starting points, different types of noise, and the effect of an impaired spectral resolution. The subjects were asked to evaluate two consecutive, differently processed sentences on the evaluation criterion listening comfort. Three different noise types were used. We asked the normal-hearing subjects to listen to the stimuli with or without simulated impaired auditory filtering. The impairment was simulated by application of spectral smearing. For the subjects with normal filtering we applied a signal-to-noise ratio (S/N) of 0 dB. For the subjects with simulated impaired frequency resolution we applied S/N=5 dB. Due to several methodological innovations (especially when the step size is adaptive), the reliability of the Simplex procedure increased. Moreover, the reliability depends on the algorithms included and on the type of noise used.

4aSC51. Talker identification from analysis of raw complex waveforms. Michael A. Stokes (MAS Enterprises, 3929 Graceland Ave., Indianapolis, IN 46208, mamodel@indy.net)

Stokes (1996) demonstrated that visual inspection of raw complex waveforms can be used to identify a vowel produced by a talker. This research resulted in the MAS Model of Vowel Perception and Production (Stokes, 1998; <http://home.indy.net/~mamodel/>). More recently, another experiment extended this work to female talkers as well as male talkers (Stokes, 2001). Together, this research represents the only ongoing comprehensive research involving visual inspection of raw complex waveforms for identifying vowels produced by any talker. As an extension of the work, the present study involves identifying a talker from a waveform display. Unique voice signatures identified from waveform displays are used to identify a talker from a set of 10 talkers in the same way as one would identify a person from fingerprints. In two trials (the word who'd in trial 1 and heed in trial 2), a talker was correctly identified from a set of 10 unique talkers per trial using small visual samples of waveforms and matching it to a waveform sample of the talkers to be identified.

4aSC52. Mapping the perceptual space defined by transition onset frequencies of synthetic CV stimuli. John W. Hawks (Kent State Univ., Kent, OH 44242, jhawks@kent.edu), Emily Wead, and Marios Fourakis (Ohio State Univ., Columbus, OH 43210)

The perception of formant transition cues to place of articulation was investigated through the use of synthetic stop-vowel stimuli. Three sets of stop-vowel stimuli were synthesized. In each set, the F_2 and F_3 onset frequencies were varied in an orthogonal design resulting in 110 combinations. The sets differed in the F_2 and F_3 transition offset frequencies which were set to be appropriate for the vowels [i, u, a]. Twenty normal-hearing listeners identified each stimulus as beginning with [b], [d] or [g]. Listeners were also asked to rate the certainty of their responses on a scale of 1 (least certain) to 5 (most certain). While resembling stop-vowel sequences, no burst or aspiration parameters were included, thus forcing listeners to use only transition information as the primary cue to place of articulation. Perceptual spaces were constructed as defined by the F_2 and F_3 onset frequencies. In each perceptual space each F_2/F_3 onset combination was assigned the phonetic code [b], [d], or [g], reflecting the plurality of subjects' responses with certainty ratings higher than 3. The resulting spaces will be discussed in terms of the effect of the target vowel on the perception of F_2 and F_3 transitions.

Session 4aUWa

Underwater Acoustics: Reverberation and Scattering

Raymond Lim, Chair

Coastal Systems Station, 6703 West Highway 98, Panama City, Florida 32407-7001

Chair's Introduction—7:40

Contributed Papers

7:45

4aUWa1. Passive sonar calibration spheres. David M. Deveau (Naval Undersea Warfare Ctr. Det. AUTECH, 801 Clematis St., West Palm Beach, FL 33401)

The need for calibrated sonar targets is addressed with the development and testing of a set of thin-walled spheres filled with a high-density fluid. Using historical research information as a guide, a set of thin-walled metal spheres was developed and filled with a high-density fluid. The combination of the spherical shape and the acoustic focusing effects of the fluid enhanced the acoustic scattering strength of the shape so that it was not only stable with temperature but also significantly greater in amplitude. The simple passive nature of the spheres makes them ideal acoustic targets for long-term unattended deployments. Using a reference level measurement, each sphere was calibrated over a wide frequency range in order to provide the user with a curve of measured scattering strength versus frequency. The resulting curves showed a high degree of correlation between the individual spheres and the modeling that was used to extrapolate the theoretical values.

8:00

4aUWa2. Scattering of plane evanescent waves by cylindrical shells and wave vector coupling conditions for exciting flexural waves. Philip L. Marston (Dept. of Phys., Washington State Univ., Pullman, WA 99164-2814)

The coupling of sound to buried targets can be associated with acoustic evanescent waves when the sea bottom is smooth. To understand the excitation of flexural waves on buried shells by acoustic evanescent waves, the partial wave series for the scattering is found for cylindrical shells at normal incidence in an unbounded medium. The formulation uses the simplifications of thin-shell dynamics. In the case of ordinary waves incident on a shell, a ray formulation is available to describe the coupling to subsonic flexural waves [P. L. Marston and N. H. Sun, *J. Acoust. Soc. Am.* **97**, 777–783 (1995)]. When the incident wave is evanescent, the distance between propagating plane wavefronts is smaller than the ordinary acoustical wavelength at the same frequency and the coupling condition for the excitation of flexural waves on shells or plates is modified. Instead of matching the flexural wave number with the propagating part of the acoustic wave number only at the coincidence frequency, a second low-frequency wave number matching condition is found for highly evanescent waves. Numerical evaluation of the modified partial-wave-series appropriate for an evanescent wave is used to investigate the low-frequency coupling of evanescent waves with flexural wave resonances of shells.

8:15

4aUWa3. Low-frequency propagation and scattering from proud targets. Gary Sammelmann (Code R22 Coastal Systems Station, 6703 W. Hwy. 98, Panama City, FL 32407-7001)

This discussion focuses on the effects of placing a large aspect ratio target in the vicinity of an interface. It uses the spheroidal T-matrix description of scattering from a large aspect ratio target developed by Hackman to describe the scattering from large aspect ratio solids. A comparison of the effects of the interface on the form function and the directivity pattern of the scattering from the target at low frequencies is made. At sufficiently low frequencies the scattering from the target and its image add coherently, giving rise to a 6-dB increase in the form function of the proud target versus a volume target. At higher frequencies the scattering oscillates between constructive and destructive interface of the scattering from the target and its image. [Work supported by Codes 321OE and 32MIW of the ONR.]

8:30

4aUWa4. Scattering by a buried elongated object. Raymond Lim (Coastal Systems Station, Code R22, 6703 W. Hwy. 98, Panama City, FL 32407, limra@ncsc.navy.mil)

The feasibility of combining spherical and spheroidal basis sets in formulating a transition matrix for scattering by elongated objects embedded in an attenuating medium such as an ocean sediment is investigated. While a description completely in terms of spheroidal bases would likely yield optimal stability, spheroidal functions are more difficult to compute and well-tested algorithms are scarce for complex arguments. Therefore, a hybrid approach that expands the surface fields of the scatterer in spheroidal bases and all other exterior fields in spherical functions is tried. This approach mitigates the need for spheroidal functions of a complex argument. The merits of the hybrid approach are assessed by comparison against standard transition matrix formulations. [Work supported by ONR.]

8:45

4aUWa5. Low-frequency propagation and scattering in the ocean. Gary Sammelmann (Code R22 Coastal Systems Station, 6703 W. Hwy. 98, Panama City, FL 32407-7001)

This discussion describes various methodologies used to model the performance of low-frequency active systems with an emphasis on developing models for low-frequency signal-to-noise estimates and coherent imaging simulations. The global matrix method of Schmidt is adapted to define a numerically stable routine for finding the solutions of the characteristic equation for the normal modes. The effects of attenuation, surface

loss, and bottom loss are added through perturbation theory. Techniques for transforming the normal-mode equations into a time-domain representation are presented. Expansions of the normal-mode functions in terms of spherical and spheroidal basis functions at the source and receiver locations are discussed. A prescription for computing reverberation is presented, and a comparison between low-frequency and high-frequency models is presented. [Work supported by Codes 321OE and 32MIW of the ONR.]

9:00

4aUWa6. Broadband ocean reverberation coherence. Jon Reeves and Gary Morella (Appl. Res. Lab., Penn State Univ., P.O. Box 30, N. Atherton St., State College, PA 16804-0030)

Utilizing a wide-band transmitting/receiving system, at-sea experimental results for broadband reverberation have been acquired for both a harsh as well as a moderate ocean acoustic environment. Both stationary and moving sonar systems were employed in the study. Replica correlation (so-called match filter) processing of linear FM (LFM) pulse codes is employed to study the degree of reverberation coherence achievable over 4 octaves of bandwidth. The LFM signals are systematically processed over a bandwidth interval from 0.5 kHz to over 12 kHz. The corresponding reverberation coherence is displayed in terms of the variance of the cross correlation coefficient as a function of time-bandwidth product. Comparison with the theoretically expected variance for band-limited white Gaussian noise is also included. [Work sponsored by L. Jacobi, ONR 333.]

9:15

4aUWa7. Solving the background mystery in acoustical resonance scattering theory (RST). Changzheng Huang (Alfred Mann Inst., Univ. of Southern California, Los Angeles, CA 90089, changzh@bmsrs.usc.edu)

The acoustical resonance scattering theory (RST) aims to solving inverse scattering problems by decomposing the backscattering echoes into a background part and a resonance part. The former contains scatterers shape information, and the latter contains the material composition information. RST assumes the existence of an intermediate background for general scatterer and surrounding combinations. However, despite the efforts taken in the 1970s through 80s, such a background still remains elusive. This long-standing difficult problem is re-investigated in this study. It is concluded here that an intermediate background or alike is not needed for the purpose of solving inverse scattering problems. The simplest rigid or soft background is sufficient for the extraction of resonance information for any type of scatterer and surrounding combinations. In other words, the missing background problem is not a problem at all.

Another new finding of this study is that a rational expression for the backscattering echoes is better suited to convey the resonance information than the additive decomposition expression as in RST. Explicit and concise relations exist between the rational expression and the acoustical properties of any scatterer, which is important for applications such as underwater target identification. This presentation will be illustrated by animation movies.

9:30

4aUWa8. The evolution of a pulse in a wave-guide and its interaction with a submerged structure: Some issues. Michael Werby (NRL, Code 7180, Stennis Space Center, MS 39529)

Initially let a pulse signal progress in time in a wave-guide along one direction *ab initio* at time t_0 . If it interacts with a submerged structure, at some time t_1 then part of the wave will be refracted and part will be reflected so that at some point along the range between the object and the source there will be interference between the pulse signal and the reflected (backscattered) signal. This, to be sure, is a function of the duration of the pulse as well as other factors. Thus, in that region we must be required to deal with a two-way solution no matter what the topography simply because of the interference between the reflected wave and the time evolving wave. Here, the most general problem is properly formulated and the effect is examined as a function of pulse width, frequency (band) width, gating, and other factors. Results are presented in the form of frames or animations and it is observed that one may gain insight into this complicated process and it is possible to make use of this effect for extracting inverse information. [Work supported by ONR and NRL.]

9:45

4aUWa9. Rough surface reverberation of surface-generated ambient noise in a nonuniform ocean waveguide. Jin-Yuan Liu, Ping-Chang Hsueh, and Shern-Hsiung Tsai (Inst. of Undersea Technol., Natl. Sun Yet-sen Univ., Kaohsiung, Taiwan, ROC)

Rough surface reverberation of surface-generated ambient noise in an ocean waveguide with a transition layer of continuously varying density and sound speed is considered in this analysis. The general formulation is based upon a theory for rough surface scattering in a stratified medium, in conjunction with the analytical solutions for wave equation in a nonuniform medium intended to simulate the sediment layer in a realistic ocean. Numerical results for various parameters for the reverberation field were obtained and analyzed. The model is intended to refine the previously proposed one [J.-Y. Liu, H. Schmidt, and W. A. Kuperman, *J. Acoust. Soc. Am.* **93**, 753–769 (1993)], and will serve as a basis for analysis of the data obtained in a recent experiment. [Work supported by NSC of ROC.]

Session 4aUWb

Underwater Acoustics: Localization and Matched-Field Processing

Kevin D. Heaney, Chair

ORINCON Corporation, 4350 North Fairfax Drive, Suite 400, Arlington, Virginia 22203

Chair's Introduction—10:10

Contributed Papers

10:15

4aUWb1. Optimal source localization in a partially saturated fluctuating waveguide. Michele Zanolin and Nicholas C. Makris (MIT, 77 Massachusetts Ave., Cambridge, MA 02139)

When the propagation path between an acoustic source and receiving array is infested with natural disturbances, such as internal waves, part of the received field may be randomized by these waveguide fluctuations. When the fluctuations cause modal decoupling sufficient for this randomized portion of the field to obey the Central Limit Theorem, it will follow a circular complex Gaussian random process in time. The total field is then said to be partially saturated. In this case the signal is partially deterministic and partially random so that both the mean and covariance of the field depend on the source position. The asymptotic bias, covariance, and signal-to-noise ratio presented in Naftali and Makris [J. Acoust. Soc. Am. **110**, 1917–1930 (2001)] for a Gaussian random field are extended to treat this more general case. Analytic expressions and simulations are performed to determine the minimum signal-to-noise ratio necessary to optimally localize an acoustic source in practical scenarios in typical continental shelf environments as a function of saturation level. The specific scenarios investigated are for the New Jersey Strataform area which serves as a natural laboratory for numerous Office of Naval Research Ocean Acoustic Experiments.

10:30

4aUWb2. Localization when the source function is known. Laurie T. Fialkowski, Joseph F. Lingeitch, Gregory J. Orris, Michael Nicholas (Naval Res. Lab., Washington, DC 20375, lfialkowski@milton.nrl.navy.mil), and John S. Perkins (Naval Res. Lab., Washington, DC 20375)

Frazer and Pecholcs [J. Acoust. Soc. Am. **88**, 995–1002 (1990)] have investigated single-hydrophone time-domain localization. They introduced several localizers, similar to an earlier method introduced by Clay [J. Acoust. Soc. Am. **81**, 660–664 (1987)], that differ in the amount of *a priori* information required about the source function. In this paper we revisit these localizers to look at the special case where the source function is known and explore the benefits of multiple hydrophones. This case is of interest due to its similarity to active matched field processing and because these localizers can be used as cost functions in environmental inverse problems when the source position is known precisely. We apply the localizers to data that includes several different types of source waveforms, and was collected in shallow water southwest of Key West, Florida. [Work supported by ONR.]

10:45

4aUWb3. Passive phase-conjugate signaling for autonomous systems in the ocean. Paul Hursky, Martin Siderius, Michael B. Porter (SAIC, 1299 Prospect St., Ste. 305, La Jolla, CA 92037), Vincent K. McDonald, and Joseph A. Rice (SPAWAR Systems Ctr., San Diego, CA 92152-5001)

Acoustic communications is an attractive way to wirelessly network distributed assets, such as UUVs, autonomous sensors, and other systems in the ocean. We have been experimenting with modulation techniques based on passive phase-conjugation (PPC), which essentially uses the

channel itself to refocus multipath arrivals, that would otherwise cause inter-symbol interference. We have also used Gold codes to isolate snapshots of the channel response, prior to the PPC operation. Our scheme can be used point-to-point, without arrays. This is a very simple demodulation technique which nevertheless is coherent and can overcome signal distortion due to multipath. Simplicity is very important, because simple algorithms consume less battery power, allowing the node to operate for a longer time in the water. We discuss details of our algorithms, including synchronization and Doppler compensation, and present the results of testing them in a variety of environments, including four SignalEx tests and the ASCOT01 experiment. We compare our PPC signaling scheme to frequency-hopped frequency-shift keying and direct sequence spread spectrum modulation using a rake receiver. All of these signaling schemes are being considered for low power, lightweight applications.

11:00

4aUWb4. Matched field processing in a highly variable shallow water site. Martin Siderius, Michael B. Porter, Paul Hursky (SAIC, 1299 Prospect St., La Jolla, CA 92037), Peter L. Nielsen (SACLANT Undersea Res. Ctr., Viale San Bartolomeo 400, 19138 La Spezia, Italy), and Jurgen Sellschopp (Forschungsanstalt für Wasserhall- und Geophysik, Klausdorfer Weg 2-24, 24148 Kiel, Germany)

Variability in the ocean environment can have a big impact on acoustic propagation. Acoustic receptions often contain multi-path contributions with fluctuations that vary significantly from the direct path to the higher order multi-path. Matched-field processing (MFP) is a beamforming method that takes advantage of the multi-path to extract information about the location of the sound source. Matched-field processing is generally successful in environments that are well known and can be incorporated into propagation models. However, in some cases the environmental conditions are too extreme for good propagation predictions. To explore the limits of MFP in highly variable environments, an experiment (ASCOT-01) was conducted last June off the New England coast. The acoustic measurements were made over several days between a sound source and a moored vertical line array of receivers. Extensive environmental measurements were made including the use of a vertical CTD chain (conductivity, temperature and depth) that was towed through the propagation path to provide a detailed view of the ocean sound speed profile. Results characterizing the difficulties of MFP at this site will be presented. Standard estimators will be compared to new alternatives intended to be robust against harsh environmental conditions.

11:15

4aUWb5. Time reversal, back propagation, matched field processors, correlation receivers, and the principles of radar/sonar signal design. Arthur Baggeroer (MIT, Rm. 5-204, Cambridge, MA 02139)

Time reversal and backpropagation have been demonstrated in several experiments. Similarly, while matched field processing (MFP) differs in terms of implementation—experimental vs computed replicas—both have

two common properties: (i) they are based on a single, spatially coherent signal; and (ii) the conjugate transpose of the Green's function and replica correlation are identical for self-adjoint systems. Hence, the principles for focusing and ambiguity plane properties of these processors are virtually identical to those for correlation receivers. The principles of optimal signal design for correlation receivers were the subject of much research for radar/sonar systems four decades ago and many of them seem to have been neglected in the analysis of time reversal, back propagation, and matched field processors. For example, time reversal from a point, a line array, or a random array of scatterers are duals of an impulse, a frequency modulated, and a pseudo-random noise signal, respectively. The equivalence and consequences of the time-bandwidth products for signals and array length wave number spread are demonstrated. The impact of sidelobes and multipath spread can be predicted. The generalizations of the important radar/sonar uncertainty principle, however, have yet been not demonstrated. This presentation reviews these optimal signal design principles and applies them to time reversal and MFP.

11:30

4aUWb6. Analytic prediction of sidelobe statistics for matched-field processing. Brian Tracey, Nigel Lee, and Lisa Zurk (Lincoln Lab., MIT, 244 Wood St., Lexington, MA 02420, btracey@ll.mit.edu)

Underwater source localization using matched-field processing (MFP) is complicated by the relatively high sidelobe levels characteristic of MFP ambiguity surfaces. An understanding of sidelobe statistics is expected to aid in designing robust detection and localization algorithms. MFP sidelobe levels are influenced by the underwater channel, array design, and mismatch between assumed and actual environmental parameters. In earlier work [J. Acoust. Soc. Am. **108**, 2645 (2000)], a statistical approach was used to derive analytic expressions for the probability distribution function of the Bartlett ambiguity surface. The distribution was shown to depend on the orthogonality of the mode shapes as sampled by the array. Extensions to a wider class of array geometries and to broadband processing will be shown. Numerical results demonstrating the accuracy of the analytic results and exploring their range of validity will be presented. Finally, analytic predictions will be compared to data from the Santa Barbara Channel experiment. [Work sponsored by DARPA under Air Force Contract F19628-00-C0002. Opinions, interpretations, conclusions, and recommendations are those of the authors and are not necessarily endorsed by the Department of Defense.]

11:45

4aUWb7. Complex spatial envelopes, invariants and matched field processing. Kevin D. Heaney and Henry Cox (ORINCON Corp., 4350 N. Fairfax Dr., Ste. 400, Arlington, VA 22203)

The waveguide invariant was introduced in the Russian Literature (S. D. Chuprov, Ocean Acoustics, 1982) to explain range-frequency interference patterns of intensity. The concept of the spatial envelope is introduced here to show how the waveguide invariant can be derived and interpreted in terms of pressure rather than intensity. It is shown how this leads to range-frequency scaling that can be used to generate finely spaced (in frequency) matched field replicas from course spaced propagation calculations. For a downward refracting sound speed profile the acoustic propagating modes can be separated into groups (refracted bottom interacting, surface-bottom interacting). These mode groups have very similar waveguide invariants. The frequency scaling can be applied to these mode groups separately. An example is presented where 150 Hz shallow water replicas are computed every 20 Hz and interpolated, yielding a factor of 30 in reduction of replica computation for MFP.

12:00

4aUWb8. Nonspecular sound reflection and phase conjugation. Leonid M. Lyamshev (Andreev Acoust. Inst., Shvernik St. 4, Moscow 117036, Russia, lyamshev@kapella.gpi.ru)

A strong back reflection is observed at certain angles of incidence of sound waves at a limited plate or shell in a liquid. This reflection is termed nonspecular reflection. It arises whenever the condition for coincidence or spatial resonance in a plate is satisfied, i.e., in other words, when the velocity of the incident wave trace coincides with the propagation velocity of any free wave in a plate or shell. If the surface of a plate or shell performs forced vibrations at the frequency doubled in comparison with the frequency of the incident wave, back reflection is observed at any incidence angle. This phenomenon is termed phase conjugation. It is caused by nonlinear interaction of an incident wave with the vibrations of a plate or shell. Fundamentals of nonspecular reflection and phase conjugation are given, i.e., theoretical concepts and some experimental data. The generality and differences of the phenomena of nonspecular reflection and phase conjugation are discussed as well as their importance for practice. [Work supported by the Russian Foundation for Basic Research.]

Session 4pAA**Architectural Acoustics and Psychological and Physiological Acoustics:
Perceptual Aspects of Real and Virtual Rooms**

Barbara G. Shinn-Cunningham, Cochair

Cognitive and Neural Systems, Boston University, 677 Beacon Street, Boston, Massachusetts 02215

Murray R. Hodgson, Cochair

*Occupational Hygiene Program, University of British Columbia, 2206 East Mall, Vancouver, British Columbia
V6T 1Z3, Canada***Chair's Introduction—1:30*****Invited Papers*****1:35****4pAA1. Challenges and solutions for realistic room simulation.** Durand R. Begault (NASA Ames Res. Ctr., M.S. 262-2, Moffett Field, CA 94035, dbegault@mail.arc.nasa.gov)

Virtual room acoustic simulation (auralization) techniques have traditionally focused on answering questions related to speech intelligibility or musical quality, typically in large volumetric spaces. More recently, auralization techniques have been found to be important for the externalization of headphone-reproduced virtual acoustic images. Although externalization can be accomplished using a minimal simulation, data indicate that realistic auralizations need to be responsive to head motion cues for accurate localization. Computational demands increase when providing for the simulation of coupled spaces, small rooms lacking meaningful reverberant decays, or reflective surfaces in outdoor environments. Auditory threshold data for both early reflections and late reverberant energy levels indicate that much of the information captured in acoustical measurements is inaudible, minimizing the intensive computational requirements of real-time auralization systems. Results are presented for early reflection thresholds as a function of azimuth angle, arrival time, and sound-source type, and reverberation thresholds as a function of reverberation time and level within 250-Hz–2-kHz octave bands. Good agreement is found between data obtained in virtual room simulations and those obtained in real rooms, allowing a strategy for minimizing computational requirements of real-time auralization systems.

2:05**4pAA2. Effects of reverberation on spatial auditory performance and spatial auditory cues.** Barbara Shinn-Cunningham (Hearing Res. Ctr., Depts. of Cognit. and Neural Systems and Biomed. Eng., Boston Univ., 677 Beacon St., Boston, MA 02215) and Norbert Kopčo (Boston Univ., Boston, MA 02215)

Ordinary room echoes and reverberation influence sound localization performance, moderately degrading judgments of source direction (i.e., judgments of azimuth and elevation) while vastly improving judgments of source distance. The relative magnitude of the influence of room acoustics on localization performance depends on the location of the listener in a room; specifically, performance is best when the listener is far from any wall (when there are no early, intense echoes). However, in addition to depending on listener location, localization performance in all three spatial dimensions depends on past experience in a particular room and improves over time. This room learning (which is not seen in a comparable anechoic study) at least partially generalizes across listener positions in a room. After reviewing results of localization studies demonstrating these effects, acoustic analyses are presented that show how echoes and reverberation alter and distort the spatial acoustic cues reaching the listener in an ordinary room and how these distortions vary with listener location in the room. Possible mechanisms to explain how spatial auditory processing might adapt in different acoustic environments in order to improve localization accuracy will be discussed. [Work supported in part by AFOSR Grant F49620-01-1-0005 and the Alfred P. Sloan Foundation.]

2:35**4pAA3. The direct-to-reverberant ratio as cue for distance perception in rooms.** Adelbert W. Bronkhorst (TNO Human Factors, P.O. Box 23, 3769 ZG Soesterberg, The Netherlands, bronkhorst@tm.tno.nl) and Pavel Zahorik (Waisman Ctr., Univ. of Wisconsin, Madison, WI 53705)

Although it is evident that in order to localize a sound source, one has to perceive both its direction and its distance, localization research has for many years focused mainly on the directional component. This situation has changed in the last few years. Recent studies have provided more insight into distance perception and the specific cues that are used by the auditory system. It appears that auditory and visual distance perception are similar, because both rely on a combination of cues, which are weighted differently in different environments. One auditory cue that has received particular attention is the direct-to-reverberant energy (D/R) ratio, because it is thought to be relatively stable, particularly in rooms. At present, however, it is not clear what the precise contribution of this cue

is or how it is coded in the auditory system. Evidence that the cue is derived from the monaural time-domain signal is contradicted by other data suggesting that it relies on binaural information. Furthermore, direct measurement of JNDs for the D/R ratio reveals unexpectedly poor performance. Merging these seemingly incompatible results into a coherent model is challenging, but necessary for obtaining a clearer picture of how the D/R cue contributes to perceived distance.

3:05

4pAA4. Investigation of reverberation synthesized by electro-acoustic enhancement systems, from a subjective and physical acoustic standpoint. Yasushi Shimizu (Architectural Acoust. Prog., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180)

Current electro-acoustic enhancement technology enables wide control over concert hall acoustics. The goal of sound field synthesis in anechoic space is to reconstruct a specific sound field. However, applying acoustic enhancement technology to existing reverberant spaces is a less developed research direction. This presentation demonstrates a methodology of electro-acoustic enhancement using regenerative reverberation through SAAF (spatially averaged acoustic feedback), an acceptable variation of RT and SPL in enhanced acoustical conditions. That was presented by YAMAHA Acoustic Research Laboratories. SAAF technology can flatten amplitude peaks at the howling frequency of acoustical feedback loops by using time variant finite impulse response filters. Therefore it enables regeneration of reverberated sound by wide band feedback in frequency without coloration. This system has been applied to “negative absorption control” and loudness equalization of under-balcony seats in current concert halls, to optimize concert hall acoustics electronically instead of architecturally. Adjusted reverberation time in enhanced condition should be between 1.5 and 2.0 times higher than the natural RT (ex. $R_{\text{on}}/R_{\text{off}}=1.8$). The SPL increases about 1 dB to 3 dB based on measured results of more than 30 performing halls integrated with acoustic enhancement system in Japan. Examples of the major Japanese concert halls with acoustic enhancement systems are presented.

3:35–3:50 Break

Contributed Papers

3:50

4pAA5. Reverberation and the Franssen effect. William M. Whitmer, William A. Yost, and Raymond H. Dye, Jr. (Parmlly Hearing Inst., Loyola Univ. Chicago, 6525 N. Sheridan Rd., Chicago, IL 60626, wwhitme@luc.edu)

The Franssen effect, in which the location of a sudden-onset (transient) tone can occlude the location of a contralateral slow-onset (steady-state) tone, has been previously shown to occur only in a reverberant space. The nature of the reverberation required—how much and what kind—for the effect to occur has yet to be determined. To explore the role of reverberation in the Franssen effect, listeners were asked to identify the location(s) of transient/steady-state pure tones with frequencies ranging from 250–4000 Hz in a variety of real and virtual contexts: (a) limited (single reflective panel) reverberant enclosure; (b) discrete reflections in an anechoic room; (c) mannequin recordings of a reverberant enclosure presented over headphones; (d) impulse-response filtered signals presented over headphones; and (e) simulated 3-D reverberation via consumer-grade software. Results indicate that the Franssen effect can be a simple, effective means to judge the verisimilitude of a virtual environment. [Work supported by NIDCD.]

4:05

4pAA6. School cafeteria noise—The impact of room acoustics and speech intelligibility on children’s voice levels. Joseph F. Bridger (Stewart Acoust. Consultants, P.O. Box 30461, Raleigh, NC 27622, joe@stewartacousticalconsultants.com)

The impact of room acoustics and speech intelligibility conditions of different school cafeterias on the voice levels of children is examined. Methods of evaluating cafeteria designs and predicting noise levels are discussed. Children are shown to modify their voice levels with changes in speech intelligibility like adults. Reverberation and signal to noise ratio are the important acoustical factors affecting speech intelligibility. Children have much more difficulty than adults in conditions where noise and reverberation are present. To evaluate the relationship of voice level and speech intelligibility, a database of real sound levels and room acoustics data was generated from measurements and data recorded during visits to

a variety of existing cafeterias under different occupancy conditions. The effects of speech intelligibility and room acoustics on childrens voice levels are demonstrated. A new method is presented for predicting speech intelligibility conditions and resulting noise levels for the design of new cafeterias and renovation of existing facilities. Measurements are provided for an existing school cafeteria before and after new room acoustics treatments were added. This will be helpful for acousticians, architects, school systems, regulatory agencies, and Parent Teacher Associations to create less noisy cafeteria environments.

4:20

4pAA7. Comparison of two speech privacy measurements, articulation index (AI) and speech privacy noise isolation class (NIC’), in open workplaces. Heakyung C. Yoon and Vivian Loftness (School of Architecture, Carnegie Mellon Univ., Margaret Morrison Carnegie Hall, Rm. 410, Pittsburgh, PA 15213, hcy@andrew.cmu.edu)

Lack of speech privacy has been reported to be the main dissatisfaction among occupants in open workplaces, according to workplace surveys. Two speech privacy measurements, Articulation Index (AI), standardized by the American National Standards Institute in 1969, and Speech Privacy Noise Isolation Class (NIC’, Noise Isolation Class Prime), adapted from Noise Isolation Class (NIC) by U. S. General Services Administration (GSA) in 1979, have been claimed as objective tools to measure speech privacy in open offices. To evaluate which of them, normal privacy for AI or satisfied privacy for NIC’, is a better tool in terms of speech privacy in a dynamic open office environment, measurements were taken in the field. AIs and NIC’s in the different partition heights and workplace configurations have been measured following ASTM E1130 (Standard Test Method for Objective Measurement of Speech Privacy in Open Offices Using Articulation Index) and GSA test PBS-C.1 (Method for the Direct Measurement of Speech-Privacy Potential (SPP) Based on Subjective Judgments) and PBS-C.2 (Public Building Service Standard Method of Test Method for the Sufficient Verification of Speech-Privacy Potential (SPP) Based on Objective Measurements Including Methods for the Rating of Functional Interzone Attenuation and NC-Background), respectively.

4p THU. PM

4pAA8. Speech intelligibility prediction of room with curved surfaces by means of its physical characteristics. Mokhtar Harun, Tharek Abdul Rahman, Md. Najib Ibrahim, and Md. Yunus Jaafar (Universiti Teknologi Malaysia, 81310 Skudai Johor, Malaysia)

In the room, it was found previously that the ear uses the first arriving sound to locate the source, but the early reflections contribute to the intelligibility. Late reflections, on the other hand reduce intelligibility. Therefore, for speech intelligibility a high sound energy fraction is necessary, requiring high early energy and low late energy. Spatial impression which is the subjective effect of early sound reflections that come from the side walls and ceiling, contributes to the sense of envelopment. This paper presents the investigation of speech intelligibility with respect to the physical characteristics of rooms with curved surfaces. The investigated physical characteristics are source-to-listener distance (L_r), listener-to-the farthest wall (L_{wmax}), listener-to-the closest wall (L_{wmin}), listener-to-the farthest wall and listener-to-the average distance to walls (L_{wave}). In addition to objective acoustics measures such as reverberation time (RT60), early decay time (EDT), average sound absorption coefficient (Aave) and clarity (C50), speech intelligibility predictors STI and Alcons were also measured and analyzed. It is the intention of this paper to explore the possibility of predicting speech intelligibility performance of a room with curved surfaces by using its physical characteristics.

4pAA9. Analysis of the uncertainty budget for the reflection and absorption coefficient given by the two-microphone method. Adolfo Esquivel D., Asoc and Guillermo Silva P. (Centro Nacional de Metrologia, km 4.5 Carr, a Los Cues, Mpio. El Marques, Queretaro 76241, Mexico)

The method for measurement of the reflection and absorption coefficient at normal incidence consists of a sound source at one end of an impedance tube, where the plane waves propagated are reflected from a sample that is at the other end of the tube. Those plane waves are generated by a random signal, and the decomposition of the standing wave pattern is carried out measuring the acoustical pressure at two-fixed microphone positions, from which it is possible to determine a transfer function to obtain the reflection and absorption coefficient. The systematic errors associated with the two-microphone technique using an impedance tube are analyzed, i.e., microphone-sample distance, microphone spacer, atmospheric measurement conditions, correction factors to the microphone sensitivity, and phase and amplitude mismatch in the microphones. These parameters constitute some of the sources to the measurement uncertainty of the measurand-reflection and absorption coefficient. The estimation of the measurement uncertainty for the reflection and absorption coefficient by the two-microphone method; likewise the analysis of the uncertainties sources and its sensitivity coefficients are discussed using a linear approach model. Furthermore, the influence of the correction factor due to the frequency response mismatch in the microphones in the calculation of the reflection and absorption coefficient is experimentally evaluated.

THURSDAY AFTERNOON, 6 JUNE 2002

LE BATEAU ROOM, 2:00 TO 4:50 P.M.

Session 4pBB

Biomedical Ultrasound/Bioresponse to Vibration: Characterization and Monitoring of Biological Flows/ Vibrational Analysis of Biological Systems

E. Carr Everbach, Cochair

Department of Engineering, Swarthmore College, 500 College Avenue, Swarthmore, Pennsylvania 19081-1397

Constantin C. Coussios, Cochair

Biomedical Engineering, University of Cincinnati, Medical Sciences Building, 231 Albert Sabir Way, Cincinnati, Ohio 45267-0761

Chair's Introduction—2:00

Invited Papers

2:05

4pBB1. Modeling strategies of ultrasound backscattering by blood. Cloutier Guy, Savery David, Fontaine Isabelle, and Teh Beng Ghee (Res. Ctr., Univ. of Montreal Hospital, 2099 Alexandre de Seve, Rm. Y-1619, Montreal, QC H2L 2W5, Canada)

Tissue characterization using ultrasound (US) scattering can allow the identification of relevant cellular biophysical information noninvasively. The characterization of the level of red blood cell (RBC) aggregation is one of the proposed applications. Different modeling strategies have been investigated by our group to better understand the mechanisms of US backscattering by blood, and to propose relevant measurable indices of aggregation. It could be hypothesized from these studies that the microstructure formed by RBC clusters is a main determinant of US backscattered power. The structure factor, which is related to the Fourier transform of the microscopic density function of RBCs, is described and used to explain the scattering behavior for different spatial arrangements of nonaggregated and aggregated RBCs. The microscopic density function was described by the Percus–Yevick approximation (nonaggregated RBCs), and for aggregated RBCs, by the Poisson distribution, the Neyman–Scott point process, and very recently by a flow-dependent rheological model. These statistical and microrheological models allowed the study of US backscattered power as a function of the hematocrit, scatterers' size,insonification frequency, and level of RBC aggregation. Experimental results available from the literature were used to validate the different approaches. [Work supported by Canadian Institutes of Health Research (MOP-36467), HSFQ, FCAR, and FRSQ.]

4pBB2. Ultrasonic scattering by blood: theories, experimental results and biomedical applications. K. Kirk Shung (Dept. of Bioengineering, 231 Hallowell Bldg., Penn State Univ., University Park, PA 16802, kksbio@enr.psu.edu)

In this paper, theoretical and experimental efforts that have been undertaken to better understand the phenomenon of ultrasonic scattering in blood will be reviewed. This subject is of interest in biology and medicine because the echoes generated by blood are used to extract blood velocity by ultrasonic Doppler flow and imaging devices. In the course of these investigations it became clear that ultrasonic scattering from blood is dependent upon such hematological and hemodynamic properties of blood as hematocrit, plasma protein concentration, flow rate and flow cycle duration. Several aspects of these experimental results have been successfully modeled by recent theoretical developments. An unexpected consequence of these efforts is that ultrasound appears to be a viable tool for blood flow visualization and hemodynamic measurements. Two unique hemodynamic phenomena that have never been reported in the hemodynamic literature have been observed: the black hole, a low echogenic zone in the center stream of whole blood flowing in a blood vessel under steady flow and the collapsing ring, an echogenic ring appearing near the periphery of a vessel at the beginning of a flow cycle, converging toward the center, and eventually collapsing during pulsatile flow. Similar observations have been made during clinical scanning of patients.

4pBB3. Blood velocity estimation by Doppler ultrasound: Problems and issues. Richard S. C. Cobbold and Aaron H. Steinman (Inst. of Biomaterials & Biomed. Eng., Univ. of Toronto, Toronto, ON M5S 3G9, Canada)

Modern methods of estimating blood flow have advanced considerably since Dr. Satamura and his colleagues at Osaka University reported the first measurements in 1959 using CW ultrasound. Most current methods involve the use of linear or phased arrays, which make possible 2-D color flow mapping, measurements from a single sample volume, and simultaneous B-mode imaging. However, along with this technology, troubling reports have been published (including our own) concerning the accuracy with which the velocity can be estimated and this has serious potential consequences in the quantitative assessment of vascular disease. The cause has not yet been completely identified, but appears to be partially associated with the wide range of Doppler angles within the sample volume, the complexity of the ultrasound propagation process, and the stochastic nature of the red blood cells flowing in the blood vessels. These causes will provide a focus for our review of the current status of Doppler ultrasound for vascular disease assessment and suggestions for future research. Specifically, a complete model of the entire measurement system is needed, and this includes the beamforming architecture, signal processing, possible nonlinear effects, scattering process, and nature of the 3-D vector flow field being measured.

Contributed Papers

4pBB4. Measurement of volumetric flow rates with ultrasonic contrast agents using a depletion method. Jeffrey A. Ketterling, Cheri X. Deng, Frederic L. Lizzi (Riverside Res. Inst., 156 William St., New York, NY 10038), Ronald H. Silverman, and D. Jackson Coleman (Weill Medical College of Cornell Univ., New York, NY 10021)

A high-intensity focused transducer was used to deplete a small, well-defined volume of contrast agent flow in a small diameter tube. The depleted volume was probed both confocally and downstream with a 7.5-MHz focused diagnostic transducer. The depletion volume was found from theoretical calculations of the acoustic beam profile. Volume flow rates were then calculated by measuring the time required for the depleted volume to pass the diagnostic transducer. Additionally, a beam with known sidelobe spacing can give information about the flow velocity if the high-intensity transducer exposure is short and the sidelobes form spatially separated subdepletion regions. A 4.7-MHz "strip electrode" high-intensity transducer with 0.5-mm sidelobe spacing was used to study the flow velocity in dialysis tubing with a 3.1-mm outer diameter. Flow of known volume velocity was generated with a syringe pump. Both Optison and Alunex contrast agents were used. Experimental flow velocities were in excellent agreement with the known flow measurement.

4pBB5. Surface response of a viscoelastic medium to subsurface acoustic sources with application to medical diagnosis. Yigit Yazicioglu, Thomas J. Royston, and Francis Loth (Univ. of Illinois at Chicago, 842 W. Taylor St., MC 251, Chicago, IL 60607, troyston@uic.edu)

The response at the surface of an isotropic viscoelastic halfspace to buried fundamental acoustic sources is studied theoretically, computationally and experimentally. Finite and infinitesimal monopole and dipole

sources are considered. Analytical and numerical integral solutions that account for compression, shear and surface wave response to the buried sources are formulated and compared with numerical finite element simulations and experimental studies on finite dimension phantom models. The focus here is on developing a better understanding of how biological soft tissue affects the transmission of vibro-acoustic energy from biological acoustic sources below the skin surface, such as a turbulent regime in an artery caused by a partial blockage. Such an understanding could catalyze the development of noninvasive procedures using an array of acoustic sensors on the skin surface for the identification and monitoring of vascular blockages a precursor to many serious cardiovascular diseases. [Research supported by NIH NCRR Grant #14250 and the Whitaker Foundation.]

4pBB6. Coupled vibration analysis of large arterial vessels. Xiaoming Zhang, Mostafa Fatemi, and James F. Greenleaf (Ultrasound Res. Lab., Mayo Clinic, Rochester, MN 55905, zhang.xiaoming@mayo.edu)

A coupled vibration model, which takes into account the elastic arterial wall, the interior blood, and exterior tissue, is presented. The arterial wall is considered as a cylindrical elastic shell and is modeled with the thick shell theory due to the large thickness-to-radius ratios. Both the blood and the tissue are considered as ideal fluids and are modeled with acoustic wave theory. A newly developed method, the wave propagation approach, is extended for the coupled vibration of arterial walls. The theoretic model is evaluated against those available in the literature. The experiments were carried out on a silicone rubber tube in a water tank with an ultrasound stimulated system. This system uses the radiation force of ultrasound to vibrate the tube at low frequency and records the resulting response by a

laser vibrometer. Both the excitation and measurement are remote and noncontact. The measured fundamental frequency is in good agreement with the theoretical prediction.

4:35

4pBB7. A USAE frequency shift method for monitoring stiffness variations in tissues. Elisa Konofagou, Kullervo Hynynen (Brigham and Women's Hospital, Harvard Med. School, Boston, MA 02115), and Jonathan Thierman (Harvard-MIT Div. of Health Sci. and Technol., Cambridge, MA)

In solid mechanics, the resonance frequency of a material is typically linked to its mechanical properties. A frequency shift therefore typically denotes a change in stiffness. It has already been shown that temperature variation causes stiffness changes in tissues. The amplitude of the

ultrasound-stimulated acoustic emission (USAE) signal has also been shown to vary with temperature as well as indicate temperature elevation. Its amplitude, however, is sensitive to both acoustical and mechanical parameters, which at most frequencies have opposite effects resulting from temperature changes. In this paper, the measurement of a frequency shift of the USAE resonant peaks is used for monitoring of the tissue stiffness variation with temperature. In a numerical simulation, the variation of the frequency shift (ranging between -150 and 500 Hz) at different temperatures is shown. Then, in a series of experiments involving a gel phantom and porcine muscle tissue, the frequency shift variation (ranging between -200 and 200 Hz) is shown to follow the known stiffness changes due to temperature. Both simulation and experimental results indicate that the USAE frequency shift method can dissociate the mechanical from the acoustical parameter dependence and detect tissue coagulation. [Work supported by NIH Grant CA82275.]

THURSDAY AFTERNOON, 6 JUNE 2002

BRIGADE ROOM, 2:00 TO 4:45 P.M.

Session 4pEA

Engineering Acoustics: Transducers, Materials and Modeling

Harold C. Robinson, Chair

NAVSEA Division, Naval Undersea Warfare Center, Code 2131, 1176 Howell Street, Newport, Rhode Island 02841-1708

Chair's Introduction—2:00

Contributed Papers

2:05

4pEA1. A synthesized tri-modal directional transducer. Alexander L. Butler, John L. Butler, Alan R. D. Curtis (Image Acoust., Inc., 97 Elm St., Cohasset, MA 02025), and Joe Rice (Naval Postgrad. School, Monterey and SSC, San Diego, CA)

A new cylindrical transducer, which achieves directional beams through the synthesis of the omni, dipole, and quadrupole extensional modes of vibration is presented. The three cylindrical radial modes are related in resonant frequency by the formula $f_n = f_0(1+n)^{1/2}$, where $n = 0, 1, \text{ and } 2$, respectively, and may be excited individually or together by the selection of the circumferential voltage distribution. It will be shown that, in the frequency band between the dipole and quadrupole modes, a simple four-step real voltage distribution can be used to synthesize a smooth transmitting response with constant beamwidth patterns. From this comparatively simple, small diameter transducer, steered beam widths of 90 deg and 10 dB front to back ratios can be obtained. Theoretical and measured transmitting response and beam pattern results on a 4.5-in.-diam. cylindrical transducer design will be presented and compared in the operating frequency range from 15 to 20 kHz. [SBIR work supported by ONR and SPAWAR.]

2:20

4pEA2. Underwater acoustic intensity measurements using a pressure/particle-acceleration intensity vector sensor. Kang Kim, Gerald C. Lauchle, and Thomas B. Gabrielson (Grad. Prog. in Acoust. and Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804)

The acoustic field of simple sources is investigated by measuring the simultaneous, collocated acoustic pressure and particle acceleration with a neutrally buoyant underwater acoustic intensity vector sensor. This sensor consists of a piezoceramic hollow-cylinder pressure transducer, and a pair of miniature accelerometers mounted inside of the cylinder. A free-field calibration is performed under the ice of a flooded quarry, while in a

laboratory test tank, the 2-D acoustic intensity field is mapped for various acoustic source configurations. These include a single spherical source and a pair of spherical sources that vibrate in or out of phase with each other. A sinusoidal signal of various frequencies between 2 and 20 kHz is gated in all of these experiments in order to suppress the effects of environmental reflections. The receiving beam pattern of the intensity sensor is also measured and reported. An analytical model for the interaction of two closely spaced spherical radiators is developed, and predictions from it compare well with the measured intensity field. Some limitations in the experiments, and future work are discussed. [Work supported by ONR, Code 321SS.]

2:35

4pEA3. The design, fabrication, and acoustic calibration of a 1-3 piezoelectric polymer composite-based cylindrical array module for towed applications. Kim C. Benjamin and Jeffrey A. Szlag (NAVSEA Div. Newport, 1171 Howell St., Newport, RI 02871)

A cylindrical transducer array module containing eight double-layered 1-3 piezocomposite elements was acoustically calibrated in the frequency range 10-100 kHz. Each wedge-shaped element, covering approximately one octant (45) of the cylindrical radiating aperture, was housed in a molded syntactic foam hub component with its outer surface conforming to a cylindrical aperture 12.7 mm in diameter. The construction approach described would allow one to connect several modules of similar (or varying) diameter along the axial direction for towed-array applications requiring directional transmit and receive performance. The double-layer element design provides two independent electrical control points for optimizing the spatial and spectral response in both transmit and receive modes. The cylindrical module fabrication approach, associated tooling, and measured acoustic results for a single module under various drive conditions will be presented to illustrate the utility of this new transducer technology. [Work sponsored by the U.S. Navy.]

2:50

4pEA4. Measurement of stress/charge coefficients of a thin polyvinylidene fluoride film. Marcellin Zahui and George Bibel (Mech. Eng., Univ. of North Dakota, Grand Forks, ND 58202)

A new experimental/analytical approach is used to measure a piezoelectric coefficient of polyvinylidene fluoride (PVDF). This method is based on the volume displacement of a vibrating beam. The obtained value of the stress/charge coefficient is checked using modal coordinate measurement theory and finite element analysis. A PVDF film is cut into two sensors shape. These sensors measure, respectively, the volume displacement and modal coordinate of a simply supported beam vibrating at its first mode. An accelerometer and a force gauge are used to measure the volume displacement and the modal coordinate of the beam. Then, PVDF volume displacement sensor equation developed by Guigou *et al.* is used with the experimental data to find the stress/charge coefficient e_{31} . This value of e_{31} is first checked using the PVDF modal sensor equation from Lee and Moon. Then, the ANSYS finite element code is used to validate the method. Experimental value of e_{31} obtained at room temperature is presented and compared with existing data. Excellent agreement is found. The extension of the method to 2D structure to determine the value of e_{31} and e_{32} simultaneously is introduced.

3:05

4pEA5. Development of a high sensitive MEMS hydrophone using PVDF. Vijay K. Varadan, Bei Zhu, and Jose K.A (212 EES Bldg., Penn State Univ., University Park, PA 16802)

The design and experimental evaluation of a PVDF-based MEMS hydrophone is presented in this paper. The basic structure of the hydrophone was fabricated on a silicon wafer using standard NMOS process technology. A MOSFET with extended gate electrode was designed as the interface circuit to the sensing material, which is a piezoelectric polymer, polyvinylidene difluoride (PVDF). Acoustic impedance possessed by this piezoelectric material provides a reasonable match to water, which makes it very attractive for underwater applications. The electrical signal generated by the PVDF film was directly coupled to the gate of the MOSFET. To minimize the parasitic capacitance underneath the PVDF film and hence improve the device sensitivity, a thick photoresist was first employed as the dielectric layer under the extended gate electrode. For underwater operation, a waterproof Rho-C rubber encapsulated the hydrophone. A silicon nitride layer passivated the active device, which is a good barrier material to most mobile ions and solvents. The device after passivation also shows a lower noise level. The theoretical model developed to predict the sensitivity of the hydrophone shows a reasonable agreement between the theory and the experiment.

3:20–3:30 Break

3:30

4pEA6. Light-weight low-frequency loudspeaker. Robert Corsaro and James Tressler (Naval Res. Lab., Washington, DC 20375, Corsaro@NRL.Navy.mil)

In an aerospace application, we require a very low-mass sound generator with good performance at low audio frequencies (i.e., 30–400 Hz). A number of device configurations have been explored using various actuation technologies. Two particularly interesting devices have been developed, both using “Thunder” transducers (Face Intl. Corp.) as the actuation component. One of these devices has the advantage of high sound output but a complex phase spectrum, while the other has somewhat lower output but a highly uniform phase. The former is particularly novel in that the actuator is coupled to a flat, compliant diaphragm supported on the edges by an inflatable tube. This results in a radiating surface with very high modal complexity. Sound pressure levels measured in the far field (25 cm) using only 200-V peak drive (one-third or its rating) were nominally 74.6 dB over the band from 38 to 330 Hz. The second device essentially

operates as a stiff low-mass piston, and is more suitable for our particular application, which is exploring the use of active controlled surface covers for reducing sound levels in payload fairing regions. [Work supported by NRL/ONR Smart Blanket program.]

3:45

4pEA7. Background noise model of the piezoresistive microphone. Allan J. Zuckerwar, Theodore R. Kuhn (NASA Langley Res. Ctr., Hampton, VA 23681), and M. Roman Serbyn (Morgan State Univ., Baltimore, MD 21231, rserbyn@morgan.edu)

A model has been developed to describe the sources of background noise in piezoresistive microphones. Three noise sources have been identified: mechanical Johnson noise due to the Brownian motion of air molecules impinging on the diaphragm, electrical Johnson noise in the bridge resistors, and generation–recombination noise in the bridge resistors. The model includes a set of lumped elements to represent the acoustic impedance of the diaphragm, another set of lumped elements to account for the stress across the bridge resistors, and a controlled source to yield an output voltage proportional to the stress due to the piezoresistive effect. Predictions based on the manufacturer’s data agree with experimental measurements. Representing the piezoresistive microphone as a nonreciprocal device, the model can be used to determine the microphone sensitivity and frequency response, as well as the background noise.

4:00

4pEA8. PC-based real-time acoustic source locator and sound capture system for teleconferencing. Ashutosh Morde, Deborah Grove, and Robert Utama (Ctr. for Adv. Information Processing, Rutgers Univ., 96 Frelinghuysen Rd., Piscataway, NJ 08854, amorde@caip.rutgers.edu)

A PC-based real time acoustic source locator and sound capture system has been developed. The system is implemented using Frontier Design A/D converters and the Intel Signal Processing Library directly on a 1 GHz Pentium III machine, without a DSP board. The source locator uses the cross-power spectral phase to locate a moving talker. The algorithm also uses an energy detector that minimizes incorrect location estimates by neglecting frames with high background noise. The source locator provides 8 location estimates per second. A 16-element 0.90 m linear delay-sum beamformer has also been implemented in the system as a method for selective sound capture. The ability of the source locator to detect talkers in a typical office environment is evaluated. In addition, the array response is measured. [Work supported by Intel.]

4:15

4pEA9. An analytical application of a time-reversal mirror in a sensor with solid cylindrical waveguides. Anthony D. Puckett and M. L. Peterson (Dept. of Mech. Eng., Univ. of Maine, 5711 Boardman Hall, Orono, ME 04469, anthony.puckett@umit.maine.edu)

A solid cylindrical waveguide or buffer rod is commonly used to isolate piezoelectric transducers from high temperature and pressure. Due to constraints on the waveguide dimensions, multiple longitudinal modes are often propagated. Dispersion and the existence of the multiple modes make it difficult to perform standard ultrasonic measurements. To simplify the measurements an analytical model was developed to model the dispersion in the multimode waveguide. This model was used to calculate the time-reversed signal required to reconstruct a pulse in the waveguide. The time-reversed signal was used in experiments to demonstrate the ability of the calculated time-reversed signal to construct a pulse. The experiments also showed the ability of the analytical model to represent the dispersion in the waveguide. Results are presented and the limitations of the analytical model presented are discussed. [Research sponsored by the Ballistic Missile Defense Organization through the ONR, Science Officer Dr. Y. D. S. Rajapakse.]

4:30

4pEA10. Implementation of CHIEF in MATLAB for prediction of sound radiated from arbitrarily shaped vibrating bodies. Mark Christensen and Koorosh Naghshineh (Mech. and Aero. Eng. Dept., Western Michigan Univ., Kalamazoo, MI 49008)

The Boundary Element Method can be used to predict sound pressure levels radiated from an arbitrarily shaped vibrating body. Using the direct, indirect, and approximation formulation one can solve such an exterior acoustic problem. In this work, the direct formulation is chosen. The two major methods employed for implementation of this formulation are the

Combined Helmholtz Integral Equation Formulation (CHIEF) and the Burton–Miller Method. It was decided that the CHIEF method would be the focus of this research. Using the CHIEF method as a guide, a computer simulation was developed that incorporates quadratic elements, calculates the sound pressure and sound power, is able to import surface velocities, and uses the program IDEAS as a preprocessor. This program was verified using theoretical models as well as experimental measurements conducted in the Western Michigan University Noise and Vibration Laboratory. This program was written in Matlab with the understanding that it can be processed on a personal computer.

THURSDAY AFTERNOON, 6 JUNE 2002

RIVERS ROOM, 1:15 TO 2:30 P.M.

Session 4pMU

Musical Acoustics and Psychological and Physiological Acoustics: Music Recognition Systems II

James W. Beauchamp, Chair

School of Music, Department of ECE, University of Illinois at Urbana-Champaign, 1114 West Nevada, Urbana, Illinois 61801

Contributed Papers

1:15

4pMU1. Independent component analysis for onset detection in piano trills. Judith C. Brown (Phys. Dept., Wellesley College, Wellesley, MA 02181 and Media Lab., MIT, Cambridge, MA 02139), Jeremy G. Todd (iZotope, Inc., Shrewsbury, MA 01545), and Paris Smaragdis (Media Lab., MIT, Cambridge, MA 02139)

The detection of onsets in piano music is difficult due to the presence of many notes simultaneously and their long decay times from pedaling. This is even more difficult for trills where the rapid note changes make it difficult to observe a decrease in amplitude for individual notes in either the temporal wave form or the time dependent Fourier components. Occasionally one note of the trill has a much lower amplitude than the other making an unambiguous determination of its onset virtually impossible. We have analyzed a number of trills from CD's of performances by Horowitz, Ashkenazy, and Goode, choosing the same trill and different performances where possible. The Fourier transform was calculated as a function of time, and the magnitude coefficients served as input for a calculation using the method of independent component analysis. In most cases this gave a more definitive determination of the onset times, as can be demonstrated graphically. For comparison identical calculations have been carried out on recordings of midi generated performances on a Yamaha Disclavier piano.

1:30

4pMU2. Two-voice fundamental frequency estimation. Alain de Cheveigné (CNRS, Ircam, 1 Pl. Igor Stravinsky, 75004 Paris, France, cheveign@ircam.fr)

An algorithm is presented that estimates the fundamental frequencies of two concurrent voices or instruments. The algorithm models each voice as a periodic function of time, and jointly estimates both periods by cancellation according to a previously proposed method [de Cheveigné and Kawahara, *Speech Commun.* **27**, 175–185 (1999)]. The new algorithm improves on the old in several respects; it allows an unrestricted search range, effectively avoids harmonic and subharmonic errors, is more accu-

rate (it uses two-dimensional parabolic interpolation), and is computationally less costly. It remains subject to unavoidable errors when periods are in certain simple ratios and the task is inherently ambiguous. The algorithm is evaluated on a small database including speech, singing voice, and instrumental sounds. It can be extended in several ways; to decide the number of voices, to handle amplitude variations, and to estimate more than two voices (at the expense of increased processing cost and decreased reliability). It makes no use of instrument models, learned or otherwise, although it could usefully be combined with such models. [Work supported by the Cognitique programme of the French Ministry of Research and Technology.]

1:45

4pMU3. Matching untrained singers using dynamic time warping. Mark Kahrs (Dept. of Elec. Eng., Univ. of Pittsburgh, 348 Benedum Hall, Pittsburgh, PA 15261)

Untrained singers lack precise control of pitch and timing, therefore any successful music search algorithm must accommodate inaccuracies in both dimensions. Furthermore, finding key signatures in untrained singers is not easy (or even plausible in some cases). Contours can be formed by up and down intervals and then matched against the database. An experimental system is described that matches an acoustic vocal input against a library of monophonic musical scores and returns a list of matched results. While partially successful, errors result from the detection of pitch on and off times as well as from pitch detection errors. Pitch detection is easiest during vowels, but the transitions from vowel to consonant can cause problems. In spite of these inaccuracies, the prototype algorithm using dynamic time warping (a dynamic programming algorithm) exceeded 95% accuracy for nine different singers for a simple tune library. However, errors computed by a difference metric can result in close misses, even for very dissimilar songs. A better metric is needed to clearly illustrate song differences.

4pMU4. Detection and representation of rhythm and tempo I: The rhythmogram. Neil P. M. Todd (Dept. of Psych., Univ. of Manchester, Manchester M13 9PL, UK)

In the paper an algorithm is presented for the computation of a primitive representation of rhythmic structure [N. P. Todd, *J. New Mus. Res.* **23**, 25–70 (1994)]. The algorithm consists of the following stages: (1) cochlear filtering, by means of an ear model based on the gammatone filter-bank [M. Cooke, *Modeling Auditory Processing and Organisation* (Cambridge University Press, Cambridge, MA, 1992)]; (2) modulation filtering, by means of a Gaussian low-pass modulation filter-bank; (3) peak detection and integration of the output of the modulation filter-bank. When the peak outputs of the modulation filters are plotted in a time-constant/time diagram they form a representation, referred to as a *rhythmogram*, which resembles a more conventional tree diagram of rhythmic structure. A single acoustic event gives rise to a single branch in the representation so that integration along a branch yields a single number corresponding to the perceptual prominence or stress of the event. The integration mechanism can be thought of as an acoustic sensory memory process and accounts for a number of important phenomena in rhythm perception, including persistence, partial backward masking, interval produced accents and grouping by temporal proximity and accent. This representation may form the input to higher level learning and recognition mechanisms.

4pMU5. Detection and representation of rhythm and tempo II: The modulation spectrogram. Neil P. M. Todd (Dept. of Psych., Univ. of Manchester, Manchester M13 9PL, UK)

In the paper a second algorithm is presented for the computation of a primitive representation of rhythmic and temporal structure [N. P. Todd, C. S. Lee, and D. J. O'Boyle, *J. New Mus. Res.* **28**, 5–28 (1999)]. The algorithm consists of the following stages: (1) cochlear filtering, by means of an ear model based on the gammatone filter-bank, (2) modulation filtering, by means of a constant- Q band-pass modulation filter-bank; (3) peak detection and sensorimotor filtering; (4) feedforward control of a simple plant to simulate tapping. The output of the modulation filter-bank produces a representation in the form of a *modulation spectrogram* whereby any periodic input gives rise to a harmonic series. Metrical structure may be inferred from the harmonic structure of the spectrum with the addition of a learning and recognition mechanism. This algorithm may account for a number of additional phenomena including the approximate Weber law property of tempo sensitivity, adaptation to perturbations in the period or phase and the autocorrelation statistics of sensorimotor synchronization. The performance of the algorithm is evaluated by comparison with the tapping behavior of 20 human subjects to the 48 fugue subjects from the Well-Tempered Clavier by J. S. Bach.

THURSDAY AFTERNOON, 6 JUNE 2002

DUQUESNE ROOM, 2:00 TO 5:00 P.M.

Session 4pNS

Noise: Noise Control Devices—Analysis and Design (Lecture/Poster Session)

Richard D. Godfrey, Chair

Owens-Corning, 2790 Columbus Road, Route 16, Granville, Ohio 43023-1200

Chair's Introduction—2:00

Contributed Papers

2:05

4pNS1. Optimally tuned vibration absorbers to control sound transmission. Michael Grissom, Ashok Belegundu, and Gary Koopmann (Penn State Univ., 157 Hammond Bldg., University Park, PA 16802, mdg184@psu.edu)

A design optimization method is proposed for controlling broadband vibration of a structure and its concomitant acoustic radiation using multiple-tuned absorbers. A computationally efficient model of a structure is developed and coupled with a nonlinear optimization search algorithm. The eigenvectors of the original structure are used as repeated basis functions in the analysis of the structural dynamic re-analysis problem. The re-analysis time for acoustic power computations is reduced by calculating and storing modal radiation resistance matrices at discrete frequencies. The matrices are then interpolated within the optimization loop for eigenvalues that fall between stored frequencies. The method is demonstrated by applying multiple-tuned vibration absorbers to an acoustically-excited composite panel. The absorber parameters are optimized with an objective of maximizing the panel's sound power transmission loss. It is shown that in some cases the optimal solution includes vibration absorbers that are tuned very closely in frequency, thus acting effectively as a broadband vibration absorber (BBVA). The numerical model and design optimization method are validated experimentally, and the BBVA is found to be an effective noise abatement tool.

2:20

4pNS2. Diffraction by jagged edge noise barriers. Jeong Ho You and Penelope Menounou (Univ. of Texas, Austin, TX 78703)

Experimental data have shown that the noise barrier's performance can be improved by introducing jaggedness into the top edge of the barrier. In the present work a combined experimental/numerical investigation was performed. The numerical investigation employs the Directive Line Source Model [Menounou *et al.*, *J. Acoust. Soc. Am.* **107**, 103–111 (2000)] and reveals the basic global characteristics of a jagged profile that increases the shielding effect over a straight edge barrier having the same average height. For a jagged edge profile consisting of finite-length segments inclined with respect to a straight top edge it was found that its performance depends on the inclination angle of the finite-length segments with respect to the horizontal, the directivity of the diffracted field, and the length of the inclined segments with respect to the spatial duration of the incident sound signal. It was also found that jagged profiles offering an increased shielding effect for certain receiver locations are different from jagged profiles offering moderate improvement over many receiver locations. The numerical predictions were verified experimentally. Experiments were carried out in air using a spark source generating N-shaped pulses, a capacitor microphone and a rigid aluminum plate with various jagged top edge profiles. [Work supported by Texas ATP.]

2:35

4pNS3. Use of tile overlays to reduce the radiation efficiency of radiating surfaces. Gary Koopmann, Eric Salesky, and Weicheng Chen (Penn State, 157 Hammond, University Park, PA 16802)

An intriguing possibility for reducing the radiation efficiency of a structure is to overlay its radiating surfaces with a contiguous set of low mass tiles. Experimental results show that significant reductions in sound power are possible depending on number of tiles, their geometry, attachment methods, mass, stiffness, and the amount of shear damping used to connect their edges to one another. This paper presents results of a combined theoretical and experimental study performed on a clamped, flat plate driven at its center and with nearly infinite baffle boundary conditions. Tiles are overlaid on the surface of the plate and lightly attached with a polymer material. A scanning laser vibrometer is used to examine the vibration patterns of the plate and the corresponding tiles for several different tile configurations over a frequency range 20–1000 Hz. Results indicate that sound power reductions occur when the independent response of each tile combines to produce an overall volume velocity that is lower than that of the plate by itself over the frequency range of interest.

2:50

4pNS4. Design and fabrication of a laboratory device for demonstration of Helmholtz resonators. Marissa Melchior, Zoren Gaspar, and Koorosh Naghshineh (Mech and Aero. Eng. Dept., Western Michigan Univ., Kalamazoo, MI 49008)

There is an increasing demand for noise emission reduction in many applications today. These applications include automotive exhausts, fan intakes and exhausts, and air conditioning ducts among many others. One way to attenuate the noise produced by these applications includes the use of a Helmholtz resonator. A Helmholtz resonator, also referred to as a side branch resonator to a duct, is simply a cavity with a known volume connected via a neck to a duct. A simple equation relates frequency and resonator geometry such that the length and cross section of the resonator neck, as well as the volume and shape of the resonator cavity, affect resonator performance. In practice a resonator is made to target a certain frequency of noise, which is to be suppressed. The noise reduction achieved is dependent on the accuracy of construction of the resonator. A laboratory setup was constructed which incorporated many of the most influential variables. This setup will be used as a demonstration device as well as a laboratory apparatus in the Mechanical Engineering curriculum at Western Michigan University.

3:05

4pNS5. In-line quincke tube muffler. William P. Patrick, Rebecca S. Bryant (United Technologies Res. Ctr.), and Larry E. Greenwald (Independent Contractor)

A unique low-pressure-drop muffler is described which has been designed to attenuate low frequency tonal noise in ducts. Flow through the muffler is divided into two noncommunicating paths in the cylindrical configuration which was designed, built, and tested. Half of the flow is ducted through a straight central annulus and the other half is ducted through a partitioned outer annulus which directs the flow in a spiral flow

pattern around the inner annulus. Thus the outer flow has a longer path length and the sound within the outer annulus is phase-delayed relative to the inner flow causing destructive interference between the inner and outer waves with resulting strong attenuation at the tuned frequencies. A procedure will be described for designing a muffler (with flow) to produce high attenuation at the fundamental noise tone and all harmonics (up to the first cross mode). Results will be presented which show that the muffler achieved over 20 dB attenuation for the first five harmonics of the incident noise in a flowing duct.

3:20

4pNS6. Jet noise from ultrahigh bypass turbofan engines. Joe W. Posey, Thomas D. Norum, Martha C. Brown (NASA Langley Res. Ctr., Hampton, VA 23681), and Thonse R. S. Bhat (Boeing Commercial Airplanes, Seattle, WA 98124-2207)

Modern commercial jet transport aircraft are powered by turbofan engines. Thrust from a turbofan engine is derived in part from the exhaust of a ducted fan, which may or may not be mixed with the core exhaust before exiting the nacelle. The historical trend has been toward ever higher bypass ratios (BPRs). The BPR is the ratio of air mass passing through the fan to that going through the core. The higher BPR engines can be more efficient and quieter. In general, a higher BPR results in lower average exhaust velocities and less jet noise. In order to address a scarcity of noise data for BPRs greater than 6, an extensive database collection effort was undertaken using the Jet Engine Simulator in NASA Langley's Low Speed Aeroacoustic Wind Tunnel. Forward flight simulations of Mach 0.1, 0.2, and 0.28 were used with BPRs of 5, 8, 11, and 14. Data was taken over the entire operating line of the simulated engines along with parametric deviations to provide a complete set of sensitivity measurements. The results will be used to develop an empirical jet noise prediction capability for ultrahigh bypass engines.

3:35

4pNS7. On windscreen insertion loss. Mariana Buzduga and Richard J. Peppin (Scantek, Inc., 7060 Oakland Mills Rd., Ste. L, Columbia, MD 21046)

Microphone windscreens are used to attenuate wind noise. They add impedance between the source and microphone, not accounted for when the system is checked by acoustical calibrator or when used in the field. The procedure for the characterization of the attenuation has recently been addressed in ANSI S1.17/2000 part 1. But to date, no windscreens have been tested in accordance with that standard. Partially, the reason is that the precision of the procedure has not been determined. As of January 2002 a round robin is addressing just that issue. By June 2002 the results of the uncertainty determination will be available. In the meantime, we have tested some of the samples used and others in a small chamber approaching free-field conditions. We present the results of the tests of insertion loss (not in accordance with S1.17 because we do not have a reverberation room). We will show that, in some cases, the use of a windscreen can easily change a measurement using type 1 instruments to type 2 or worse. Without knowing information about a particular windscreen, the use of a windscreen in still air can drastically change uncertainty of measurement. In moving air conditions we expect even more severe problems.

Poster Papers

All posters will be on display from 1:30 p.m. to 5:00 p.m. Contributors will be at their posters from 3:50 p.m. to 5:15 p.m. To allow for extended viewing time posters will remain on display until 10:00 p.m.

4pNS8. Automation of calculation and design of means of protection of buildings from transport noise. Igor L. Shubin and Irina L. Sorokina (Dept. of Noise Control, Res. Inst. of Bldg. Phys. (NIISF), 21 Lokomotivny proezd, Moscow, 128239 Russia)

There are stated basic principles of design using the created computer program on calculation and projection of means of protection of the build-

ings from transport noise. A number of problems have been considered which are connected with the use of the existing methods of evaluation of the conditions of transport noise distribution in the urban area and the effectiveness of protection of buildings and constructions from noise (with the use of the automated calculations and design of noise protection).

4pNS9. Noise reduction of a table saw. John Carlson (School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

The National Institute for Occupational Safety and Health (NIOSH) is sponsoring a design project to address the noise levels that commonly exist at construction worksites. Through engineering control, the problem of noise emission from a table saw will be addressed. The noise emitting sources will be pinpointed using a sound pressure level meter. With this knowledge, the next step will be to reduce the sound pressure levels at the noise sources. This will be done by using noise reduction techniques such as insulation, and vibration dampening. The goal is to reduce the noise emission to a level between 85 and 90 dB(A).

4pNS10. Noise source identification and control of a contractor grade table saw. Kristin Bleedorn, Matthew McKee, Dale Yarbough, Jr., Chen Yu, Edward L. Zechmann, and J. Adin Mann III (Aero Eng. and Eng. Mech., Iowa State Univ., 2271 Howe Hall, Ames, IA 50011)

Sponsored by the National Institute for Occupational Safety and Health (NIOSH) as part of their initiative to explore noise reduction strategies for construction equipment, a team of engineering students at Iowa State University studied a contractor grade table saw. Based on standards, published work, and preliminary tests, a repeatable noise measurement procedure was developed for the table saw operation. The wood-feed rate and force were measured. With the saw operating in a standard and consistent manner, noise sources on the saw were identified using sound intensity measurement techniques and through the application of noise control strategies to individual sources. At this stage, noise control strategies, such as enclosing the motor, are effective for noise source identification but not practical. The effectiveness of both approaches to identifying the noise sources will be discussed. Based on rank ordering the contribution of each noise source to the overall sound levels, permanent noise control strategies are suggested.

4pNS11. Noise control of a circular saw. Jeff Okrzesik (School of Mech. Eng., Purdue Univ., West Lafayette, IN 47907)

Construction workers are subjected to high levels of noise pollution on a daily basis. Sound levels of about 85 dB(A) with exposures of 8 hours per day will result in permanent hearing loss after many years [Noise and Hearing Loss, NIH Consensus Statement Online 8(1), 1–24 (1990)]. There are numerous sources contributing to the noise problems. Tools are of particular interest considering the amount of time the average worker uses them. A circular saw is a common tool used daily by many construction workers. The levels of noise produced by a circular saw will be tested in order to evaluate the problem. The noise levels as well as the locations of the noise sources will be evaluated. Several ideas regarding the reduction of sound emissions have been researched. There will be three initial focal points of the investigation. The blade properties, including the material, size and tooth design, are all key elements that will be studied closely. The numerous dynamic parts in the motor are also large sources of noise that may be reduced by increasing the efficiency of the motor. Another vital element to reducing noise involved with a circular saw is to study the interaction noise from the piece being worked on.

4pNS12. Aeroacoustics of microphones. Daniel Finfer (Purdue Univ., 1077 Ray W. Herrick Labs., West Lafayette, IN 47907-1077, finfer@ecn.purdue.edu)

Several commercial options exist for prevention of the pseudo-sound often introduced by a microphone in the presence of a flow field. One such device is a so-called “slit-tube.” A slit-tube is a tube approximately the diameter of the microphone with which it is used. The length of such a tube can vary anywhere from 25 cm to 50 cm, and it is most generally fit with a stream-lined noise cone. In theory, a perforated slit along the length of the tube encourages net cancellation of turbulent pressure variations, while not interfering with the acoustic field. This poster compares the effectiveness of two similar slit-tube designs. Each tube was placed inside of a wind-tunnel and exposed to white noise. The effects of increasing flow velocities on the transfer-function between the source and receiver signals are analyzed.

THURSDAY AFTERNOON, 6 JUNE 2002

STERLINGS ROOMS 2 AND 3, 1:00 TO 2:15 P.M.

Session 4pAa

Physical Acoustics: General Acoustics Modeling

Doru Velea, Cochair

Planning Systems, Incorporated, 12030 Sunrise Valley Drive, Reston Plaza I, Suite 400, Reston, Virginia 20191

Roger M. Waxler, Cochair

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

Contributed Papers

1:00

4pAa1. Integral equation approach for analyzing acoustic interactions with wrinkled flame fronts. Lei Wu and Tim Lieuwen (School of Aerosp. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0150, lei.wu@ae.gatech.edu)

Flame-acoustic wave interactions play an important role in the characteristic unsteadiness of turbulent combustion processes. They arise because acoustic waves incident from other parts of the flame or from ex-

ternal sources impinge upon the flame and are scattered because of the significant change in sound speed and density at the flame front. The characteristics of the scattered waves are complex, due to the fact that they are interacting with a dynamic flame surface that is convoluted over a broad range of length and time scales. Over a wide range of frequencies (e.g., up to about 20 kHz), such turbulent flames may be treated as non-passive (i.e., the flame acts as a source of acoustic energy) temperature discontinuities. We describe here an integral equation approach, similar to the development of the Ffowcs Williams–Hawking (FWH) equation, that

is suitable for analysis of acoustic-flame interactions. This FWH equation is generalized for the case of interest where the surface is penetrable, not passive, and whose area varies in time. The integration surface consists of the time varying flame front that divides two homogeneous regions. The wave motions in these two regions are coupled by appropriate mass, momentum, and energy conservation conditions.

1:15

4pPAa2. New anisotropic perfectly matched layer boundary condition for elastic waves in solids. Yibing Zheng (Physical Acoust. Lab., Dept. of Mech. Eng., Yale Univ., New Haven, CT 06520-8286), Xiaojun Huang (MIT, Cambridge, MA 02142), and Robert E. Apfel (Yale Univ., New Haven, CT 06520-8286)

The perfectly matched layer (PML) is currently a popular absorbing boundary condition for numerical modeling of electromagnetic, acoustic, and elastic wave problems. Existing PML formulation for elastic waves uses a field splitting scheme, which increases the number of variables in the computation. The new anisotropic elastic wave PML modifies the elasticity or compliance tensors to be anisotropic. It utilizes fewer variables for both isotropic and anisotropic media, so that computing memory and time can be reduced significantly. Its formulations for Cartesian and curvilinear coordinate systems will be discussed. Moreover, this anisotropic PML can be directly used in the finite element method, and easily be implemented in the finite difference time domain method as well. [Work supported by NASA (NAG3-2147) and by NSF (CTS-9870015).]

1:30

4pPAa3. A simple analytical model for prediction of aeroacoustic cavity oscillations. Donald Bliss and Linda Franzoni (Dept. of Mech. Eng., Duke Univ., Durham, NC 27708-0300)

High-speed flow over cavities in vehicle surfaces often produces intense tonal pressure fluctuations. Although this problem has been of concern for decades, especially for aircraft, the underlying physical mechanisms have not been well understood. A fairly simple analytical model has been developed that improves understanding of the physical mechanisms of pressure fluctuations in shallow rectangular cavities exposed to high-speed flow. The new model describes the important features of the phenomena, and gives good agreement with experimental data. The analysis is constructed by combining waves that can exist in a system having a finite thickness shear layer dividing two acoustic media, one at rest and one in motion. The wave types for this configuration can be interpreted in physical terms. A new aspect of the analysis shows the important effect of shear layer thickness on the convective waves. Fundamentally, only two convective waves and three acoustic waves suffice to represent the phenomenon. The analysis shows the important role of mass addition and removal at the rear of the cavity due to shear layer oscillation. Certain conditions on energetics and shear layer motion at the rear of the cavity must be satisfied simultaneously for oscillation to occur.

1:45

4pPAa4. On the effect of viscosity in scattering from partially coated infinite cylinders. Jerry H. Ginsberg (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405)

Partial coating of a submerged object introduces discontinuities in the surface properties that complicate an acoustic scattering analysis. Some analyses, such as the work by Feit and Cuschieri [J. Acoust. Soc. Am. **92**, 1721–1728 (2000)], have used weighted residual techniques based on global series representations, thereby generating a solution that satisfies discontinuous boundary conditions in an average sense. Others, such as Ferri, Ginsberg, and Rogers [J. Acoust. Soc. Am. **92**, 1721–1728 (1992)] have used collocation-type formulations. Another issue arises from the fact that the normal velocity changes discontinuously at an impedance discontinuity. The associated large velocity gradients give rise to concern that viscous shear might play a role. The present work considers the two-dimensional problem of scattering of a plane wave incident on an infinite cylinder that is coated with strips of pressure-release material extending over quadrants on the illuminated and shadowed sides, with the remainder of the surface considered to be rigid. The analysis represents the scattered field in terms of acoustic and vortical contributions, both of which are represented in series form. Solutions obtained by the method of weighted residuals are shown to be less accurate than those obtained from a collocation procedure. The results for surface pressure and farfield directivity indicate that viscous effects are important only if the Reynolds number is extremely small.

2:00

4pPAa5. Acoustic field in a cased well with a sectorial crossing channel. Weijun Lin, Chengyu Zhang, Xiuming Wang, and Hailan Zhang (Inst. of Acoust., Chinese Acad. of Sci., P.O. Box 2712, Beijing 100080, PROC, weijun_1@hotmail.com)

The crossing channel in a cased well is a vertical and sectorial column filled with fluid in the cement layer between the steel pipe and formation sometimes formed during well cementing, which is harmful to the well and must be detected. To study the possibility of detecting the crossing channel by using a traditional logging tool, an acoustic field generated by a monopole source in a cased well with a crossing channel of various angles is simulated by a 2.5-D finite difference method for the first time. The results of the 2.5-D finite-difference method for the cases of 0 or 360 deg are compared and consistent with that of the well-known real axis integration method. Snapshots of normal stress and synthetic time-domain waveforms are displayed. The 2-D spectrum in wave number and frequency domain is also calculated, where the influence of the channel is clearer than that in the waveforms. Numerical study demonstrated that a crossing channel can be detected and sized if its angle is greater than 30 deg, and larger offset and lower frequency source are favorable to detect and size the crossing channel. [Work supported by NSFC 19974059.]

Session 4pPAb

Physical Acoustics and Education in Acoustics: Demonstrations in Physical Acoustics

Matthew E. Poese, Chair

Graduate Program in Acoustics, Pennsylvania State University, P.O. Box 30, State College, Pennsylvania 16801

The Technical Committee on Physical Acoustics and the Committee on Education in Acoustics are proud to present a showcase of demonstration apparatus that promises to inspire and challenge your understanding of acoustics. Among the demonstrations are the following:

Acoustic Log Starter: 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.

Longitudinal Waves on a String: A "Long Stringed Instrument" will be played. The instrument generates interesting sounds when the steel piano wire strings are excited with longitudinal waves instead of the usual transverse waves.

High Amplitude Waves on a String: Nonlinear waves on a string show "mode jumping effects." The string is a very stretchy light elastic cord and the amplitude can be made quite large.

Standing Waves on a Drum Head: A thin latex sheet is stretched on a circular frame and excited into motion by a loudspeaker placed below the membrane. Beautiful standing wave shapes are stroboscopically illuminated when the drive frequency of the loudspeaker is coincident with a mode of the membrane.

Listening to Bubbles: Ambient noise in the ocean is due in part to bubbles. Small, individual bubbles are entrained when rain drops hit the ocean surface and clouds of bubbles are created under breaking waves. The unique sounds of these events will be demonstrated.

Singing Beer Glass: Bubbles can greatly affect the frequency dependent speed and attenuation of sound. This will be demonstrated using a tube full of bubbles and a glass of beer.

Cavitation Stick: Collapsing vapor bubbles are responsible for damage to pumps and propellers in a process known as cavitation. The phenomenon will be demonstrated in dramatic fashion using a simple device called a "cavitation stick."

Helmholtz Resonator Jets: Not only can a Helmholtz resonator provide a jet with enough force to blow out a match, but it can also provide the motive force for an acoustic motor. Four Helmholtz resonators arrayed on a crossbar and balanced at the center of the cross are made to spin around the balance point by creating a high-intensity, two-dimensional standing wave field near the resonators.

Spinning Cups with Sound: 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.

Acoustic Levitation: Thin sheets of foam are levitated in an intense standing wave field. This effect can be explained with the Bernoulli principle due to large particle velocities at the pressure nodes of the standing wave. Oscillation of the trapped material about its levitation position exhibits an interesting instability.

Acoustic Fountain: A horizontal tube half-filled with water displays miniature fountains when excited with a standing wave. The fountains appear at pressure nodes due to the low-pressure cells caused by the radiation pressure at the pressure anti-nodes.

Singing Flames: An empty tube can be made to resonate at its fundamental frequency when suspended over a flame from a Bunsen burner.

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.

Acoustic Land Mine Detection: A demonstration of the process of imaging buried objects will be shown.

The collaborators for this show, in alphabetical order are:

Anthony A. Atchley, Kevin J. Bastyr, Steven L. Garrett, Robert M. Keolian, Murray S. Korman, Julian D. Maynard, John MacGillivray, Matthew E. Poese, James M. Sabatier, Charles R. Thomas, Ray Scott Wakeland, Preston S. Wilson

Session 4pSA**Structural Acoustics and Vibration and Noise: Active Control of Ducted Fan Noise and Other Active Control Issues**

Scott D. Sommerfeldt, Chair

*Department of Physics and Astronomy, Brigham Young University, P.O. Box 24673, Provo, Utah 84602-4673***Chair's Introduction—2:00*****Invited Papers*****2:05**

4pSA1. Error signal sensing for active noise control in turbofan engine nacelles. Bruce Walker, Alan Hersh (Hersh Walker Acoust., 780 Lakefield Rd., Unit G, Westlake Village, CA 91361), and Edward Rice (E. J. Rice Consulting, 3116 Dover Ctr. Rd., Westlake, OH 44145)

Blade passage harmonic tones are a significant component of turbofan engine noise under approach conditions. Active noise control (ANC) offers a tool for suppression of these tones, with the caveat that interference patterns in the sound field can lead to poor or even negative correlation between in-duct sound pressures and radiated sound power. Various modal decomposition and wave-number separation schemes have been investigated in attempt to overcome this obstacle. The paper will discuss methods that have been applied to design and implementation of microphone arrays and signal processing techniques to provide unambiguous error signals for ANC systems in these complex acoustical environments and will present results of computer simulations and field tests. [Work sponsored by NASA Glenn and NASA Langley Research Centers.]

2:30

4pSA2. Numerical model for multi-mode active control of turbofan noise. Laralee G. Ireland and Scott D. Sommerfeldt (Dept. of Phys. and Astron., Brigham Young Univ., Provo, UT 84602, ireland@byu.edu)

A numerical model has been developed to investigate the possibility of implementing active control to minimize noise radiation from turbofan engines. Previous experimental work on the NASA Glenn Research Center ANC fan have been encouraging, but the question remains of whether the modal approaches investigated can be expected to work effectively on real engines. The engine model presented here uses an indirect boundary element method, implemented with Sysnoise, and a multi-mode Newton algorithm to simulate the active control. Experimental results obtained using the NASA Glenn ANC fan, which has a simplified cylindrical engine geometry, were used as a benchmark for the model. The model results have been found to compare well with the experimental results. Recently, a more realistic, but still axisymmetric, engine geometry has been used. Preliminary results will be presented which indicate that a modal ANC approach should still be effective for controlling turbofan noise.

2:55

4pSA3. Fan noise control using Herschel–Quincke resonators on a production turbofan engine. Ricardo A. Burdisso (Vib. and Acoust. Labs., Virginia Tech, 153 Durham Hall, Blacksburg, VA 24061) and Carl H. Gerhold (NASA Langley Res. Ctr., Hampton, VA 23681-2199)

The Herschel–Quincke (HQ) resonator concept is an innovative technique that consists of installing circumferential arrays of HQ waveguides around the inlet of a turbofan engine. An HQ waveguide is essentially a hollow side tube that travels along (but not necessarily parallel to) the engine axis and attaches to the inlet at each of the two ends of the tube. To investigate the potential of the concept, the approach was tested on a full-scale production Honeywell TFE731-60 engine. An HQ–inlet system containing two arrays was designed to attenuate the blade passage frequency (BPF) tone at approach condition, i.e., 60% engine power. However, the system was tested over the full range of engine power settings. The effects of each array both individually and together were evaluated as compared to the hard-wall case. Both far-field and induct data were recorded during the tests. The results show good attenuation of both the BPF tone and broadband components. Furthermore, reduction of “buzz-saw” tones, i.e., additional tones radiated from the inlet when the fan-tip speed goes supersonic, was observed with the HQ system. Some fan distortion effects and increase in noise was observed at higher engine speeds. [Work supported by NASA Langley Research Center.]

3:20–3:30 Break

3:30

4pSA4. Development of shaped PVDF sensors for the measurement of volume displacement. Brian Zellers, Randall Rozema, and Koorosh Naghshineh (Mech. and Aeronautical Eng. Dept., Western Michigan Univ., Kalamazoo, MI 49008)

To attenuate the sound radiated from a vibrating structure, active control can be implemented. When this radiated sound is in the lower frequencies, a reduction of the surface volume displacement of the vibrating structure is suitable for achieving attenuation within these lower frequencies. The sound power emitted from the surface is directly related to the surface volume displacement of the vibrating structure. Traditionally, measurement of the vibration of the structural surface is done using multiple point sensors. Previous work has shown that for a one-dimensional vibrating structure a single sensor made of polyvinylidene fluoride (PVDF) can be used in place of point sensors. Additionally, PVDF sensors have been developed to sense the surface volume displacement in a local area of interest by spanning only that area. However, it has been observed that at higher frequencies both the total and local sensors tend to exhibit reduced accuracy. Thus, a series of experiments was conducted to determine if the PVDF stress/charge coefficient varies as a function of frequency. Additionally, a simulation program was developed to predict sensor charge output for arbitrarily shaped sensors. This simulation, which allows us to develop sensor shapes and predict sensor placement errors, was verified experimentally.

3:45

4pSA5. Decentralized structural acoustic control of a launch vehicle payload fairing. Kenneth D. Frampton (Dept. of Mech. Eng., Vanderbilt Univ., VU Station B 351592, Nashville, TN 37235)

The development of smart structures and active noise and vibration control technologies promised to revolutionize the design, construction and, most importantly, the performance of many complex engineering. However, the early promise of these technologies has not been realized in large-scale systems primarily because of the excessive complexity, cost, and weight associated with centralized control systems. Now, recent developments in MEMS sensors and actuators, along with networked embedded processor technology, have opened new research avenues in decentralized controls. Such a control system consists of numerous nodes, possessing limited computational capability, sensors, and actuators. Each of these nodes is also capable of communicating with other nodes via a wired or wireless network. This results in a dramatic shift in the control system paradigm from that of a single, centralized computer to that of numerous decentralized, networked processors. This work describes the application of such a control system to the reduction of structural acoustic radiation in a launch vehicle payload fairing. A JAVA-based simulation tool is employed to simulate the interactions of the physical system with the networked embedded controllers. Results will indicate the potential for such a control system as well as the limitations imposed by the networked embedded processor hardware.

4:00

4pSA6. A comparison of variable structure control strategy to classical control strategies for active vibration control of distributed structures. Jiaqiang Pan (Dept. of Instrumental Sci. & Eng., Zhejiang Univ., Hangzhou 310027, PROC) and Shuwen Pan (Univ. of California, Irvine, CA 92697)

Independent Modal Space Control (IMSC) is known as an essential method of active vibration control of distributed elastical structures. However, Meirovitch, who advanced the method at the beginning of the 1980s, and other researchers have only investigated modal control law and control performance due to employing classical linear control strategies (e.g., pole allocation, optimal control, etc.) since the 1980s. In this study, au-

thors theoretically and experimentally investigate the modal control law and control performance as using a nonlinear control strategy called variable structure control (VSC) in IMSC of the distributed structures, and then give a comparison of them to the results from using classical control strategies. Besides strong stability robustness of VSC, the theoretical analysis also shows that, when the sensor-actuator couples employed are less than the number of the modes one desires to control, use of VSC can efficiently avoid the control spillover that often happens at residual modes. For active vibration control of the elastical structures with the classical control strategies, the control spillover usually is a difficult problem to deal with. [Work supported by Zhejiang STPF, PROC.]

4:15

4pSA7. Multi-channel active control of axial cooling fan noise. Kent L. Gee and Scott D. Sommerfeldt (Dept. of Phys. and Astron., N-283 ESC, Brigham Young Univ., Provo, UT 84602, klg33@email.byu.edu)

A multi-channel active control system has been applied to the reduction of free-field tonal noise from a small axial cooling fan. The experimental apparatus consists of an aluminum enclosure which houses the fan, an infrared detector-emitter pair which serves as a reference sensor, loudspeakers, microphones, and appropriate filters and amplifiers. The research fan and loudspeakers have been modeled theoretically as point sources to derive a mathematical expression for radiated power. The minimization of this power has served to guide the number and location of control sources, as well as to discover potential error microphone locations in the extreme near-field. Experiments with these microphone positions have shown that there are predictable near-field locations which consistently lead to significant reductions in the global mean-squared pressure for the first four harmonics of the blade passage frequency. For example, a four channel configuration with the error microphones placed appropriately results in global mean-squared pressure reductions on the order of 10 dB or more for the targeted frequencies.

4:30

4pSA8. Design of model reduction and robust controller for underwater echo cancellation. Woosuk Chang, Vijay K. Varadan, and Vasundara V. Varadan (212 EES Bldg., Penn State Univ., University Park, PA 16802)

This paper presents the design and development of a simple and robust control system for structural acoustics based on state space control theory. The controller was developed using model Hankel singular value (MHSV)-based model reduction, and analog controllers are fabricated. MHSV is developed for a model coordinate system describing vibrating structures, and analog controllers were designed from transfer functions of controllers, which have great advantage over digital controllers in terms of simplicity and cost. This controller design for structural acoustics is applied for controlling sound transmission and reflection. MHSV shows good model reduction of finite-element-based structure models. The feasibility of this controller for the underwater active reflection cancellation is also studied. 1-3 piezocomposite material is used as a transducer that generates the acoustic signal. Its electroacoustic model is developed using the Mason circuit and the control parameters are directly derived. The controller has been designed accordingly and its performance is evaluated using pulse tube. The simulation and measurement show the possibility of achieving a very good reflection reduction in a desired frequency band.

Session 4pSC**Speech Communication and Psychological and Physiological Acoustics: Future of Infant Speech Perception Research: Session in Memory of Peter Jusczyk**

Rochelle S. Newman, Cochair

Department of Hearing and Speech Science, University of Maryland, College Park, Maryland 20742

Ruth J. Tincoff, Cochair

*Psychology Department, Harvard University, 970 William James Hall, 33 Kirkland Street, Cambridge, Massachusetts 02138***Chair's Introduction—1:00*****Invited Papers*****1:05****4pSC1. Word recognition and phonetic structure acquisition: Possible relations.** James Morgan (Cognit. & Linguistic Sci., Brown Univ., Box 1978, Providence, RI 02912)

Several accounts of possible relations between the emergence of the mental lexicon and acquisition of native language phonological structure have been propounded. In one view, acquisition of word meanings guides infants' attention toward those contrasts that are linguistically significant in their language. In the opposing view, native language phonological categories may be acquired from statistical patterns of input speech, prior to and independent of learning at the lexical level. Here, a more interactive account will be presented, in which phonological structure is modeled as emerging consequentially from the self-organization of perceptual space underlying word recognition. A key prediction of this model is that early native language phonological categories will be highly context specific. Data bearing on this prediction will be presented which provide clues to the nature of infants' statistical analysis of input.

1:25**4pSC2. The future of infant speech perception research: Gotta wear shades.** Amanda C. Walley (Dept. of Psychol., Univ. of Alabama, Birmingham, Birmingham, AL 35294)

The future of infant speech perception research is very bright in large part because of the legacy left to us by Peter W. Jusczyk. I will selectively review a number of studies, many of them conducted by Peter and his colleagues, that I consider to be especially interesting and/or provocative and that have helped to advance our understanding of the development of speech perception. For example, in two fairly recent studies, 4.5 to 6-month-old infants were shown to display some incipient spoken word recognition ability [D. R. Mandel *et al.*, *Psychol. Sci.* **6**, 314–317 (1995); R. Tincoff and P. W. Jusczyk, *ibid.* **10**, 172–175 (1999)]. In effect, these studies have smashed the sound-meaning barrier that was, according to traditional estimates, placed somewhere between 9 and 13 months of age and have important implications for current developmental theory. This work is also illustrative of one trend apparent in recent research, namely, to accord greater attention to infants' perception of ecologically-relevant stimuli and/or their perception of speech stimuli in ecologically-relevant contexts. Still, there is much to be learned about spoken word recognition beyond infancy and before adulthood if we are to arrive at a complete picture of this core aspect of spoken language processing.

1:45**4pSC3. From speech perception to word learning and beyond.** Janet F. Werker (Dept. of Psych., 2136 West Mall, Univ. of British Columbia, Vancouver, BC V6T 1Z4, Canada)

From his undergraduate days at Brown, Peter Jusczyk revolutionized our understanding of the link between infant speech perception and language use. This talk reviews how Jusczyk's work influenced my research in three important domains. The talk begins with a discussion of Jusczyk's early work on infant speech perception, and illustrates how that led to my initial work on infant cross-language perception. This is followed by a discussion of Jusczyk's work on sensitivity to probabilistic information and how this influenced the more recent work [Maye, Werker, and Gerken] on the mechanisms underlying changes in phonetic perception. The third research section briefly describes how Jusczyk's work on word segmentation influenced ongoing research in the laboratory on early word learning. In the final few minutes of the talk, an attempt will be made to give a sense of the enormous influence Peter Jusczyk's work had, and continues to have, on our field.

4pSC4. The development of perceptual attention and articulatory skill in one or two languages. Carol Fowler and Catherine Best (Haskins Labs., 270 Crown St., New Haven, CT 06511)

Infants acquire properties of their native language especially during the second half of the first year of life. Models, such as Jusczyk's WRAPSA, Best's PAM, Kuhl's NLM, and Werker's account describe changes in perceptual or attentional space that may underlie the perceptual changes that infants exhibit. Unknown is the relation of these changes to changes in speechlike vocalizations that occur at the same time. Future research should address whether the perceptual models predict production learning. Other issues concern how the perceptual and articulatory systems develop for infants exposed to more than one language. Do multiple perceptual spaces develop, or does one space accommodate both languages? For infants exposed to just one language, but living in an environment where the ambient and pedagogical language is different (say, infants in a monolingual Spanish home in the U.S.), early language learning fosters learning the native language, but it may impede learning the ambient language. How much or how little does early exposure to the ambient language allow development of perceptual and articulatory systems for the ambient language? A final issue addresses whether the emergence of lexical, morphological and/or syntactic abilities in the second year is related to further changes in speech perception and production. [Work supported by NICHD.]

2:25–2:50

Discussion

Sandra Trehub, Discussion leader

2:50–3:00 Break

Contributed Papers

3:00

4pSC5. Talker variation and word learning. George Hollich III, Peter Jusczyk (Johns Hopkins Univ., 3400 N. Charles St., Baltimore, MD 21218), and Michael Brent (Washington Univ.)

While infants must go beyond talker-specific information in recognizing a given word, regardless of the talker, they must also process talker-specific information in order to extract meaning from a particular sound source. Otherwise, for example, they could never recognize whether [hct] referred to a talker's pronunciation of hot, hut, or even hat. This poster suggests that not only do infants process talker-specific information, but they also make use of it both to extract invariant properties in learning a new word and in recognizing talker-specific tokens faster. Using the splitscreen preferential looking paradigm, two studies were conducted that examined how talker-specific properties and variation among talkers could facilitate word learning. Results of study 1 indicated that word learning was facilitated in the case where infants heard different talkers. Thus, talker variation is critical for the extraction of invariant properties of a word. However, the results of study 2 indicated that talker-specific properties were encoded and used to help these infants recognize and learn the referents of these words. Given this evidence, it is suggested that infants appear to be using talker-specific information to form abstract representations of the invariant properties of words.

3:15

4pSC6. The role of language experience in word segmentation: A comparison of English, French, and bilingual infants. Linda Polka, Megha Sundara, and Stephanie Blue (School of Commun. Sci. and Disord., McGill Univ., Montreal, QC H3G 1A8, Canada)

English-learning and French-learning 7.5-month-old infants were tested using the headturn preference procedure on their ability to segment bisyllabic words in both English and French. In the English condition infants were familiarized with trochaic bisyllables, the predominant stress pattern found in English, and then presented English passages with and without the familiarized words. In the French condition, infants were familiarized with iambic bisyllables, the characteristic word stress pattern found in French, and were then presented French passages with and without the familiarized words. Findings indicate that by 7.5 months of age,

infants' learning either a syllable-timed (French) or a stressed-timed language (English) can segment bisyllabic words with the predominant stress pattern of their native language. However, French infants fail to segment English trochaic words from English passages; data on English infants' segmentation in the French condition are forthcoming. If both groups fail to segment in a rhythmically different non-native language, it will confirm that word segmentation abilities of 7.5-month-old infants are influenced by the prosodic structure of the native language. Preliminary results obtained from infants who are regularly exposed to both languages will also be reported.

3:30

4pSC7. Language discrimination without language: Experiments on tamarin monkeys. Ruth Tincoff, Marc Hauser, Geertrui Spaepen (Dept. of Psych., Harvard Univ., 33 Kirkland St., Cambridge, MA 02138, tincoff@wjh.harvard.edu), Fritz Tsao (Hillsdale College, Hillsdale, MI 49242), and Jacques Mehler (LSCP, 75006 Paris, France)

Human newborns can discriminate spoken languages differing on prosodic characteristics such as the timing of rhythmic units [T. Nazzi *et al.*, *JEP:HPP* **24**, 756–766 (1998)]. Cotton-top tamarins have also demonstrated a similar ability to discriminate a morae- (Japanese) vs a stress-timed (Dutch) language [F. Ramus *et al.*, *Science* **288**, 349–351 (2000)]. The finding that tamarins succeed in this task when either natural or synthesized utterances are played in a forward direction, but fail on backward utterances which disrupt the rhythmic cues, suggests that sensitivity to language rhythm may rely on general processes of the primate auditory system. However, the rhythm hypothesis also predicts that tamarins would fail to discriminate languages from the same rhythm class, such as English and Dutch. To assess the robustness of this ability, tamarins were tested on a different-rhythm-class distinction, Polish vs Japanese, and a new same-rhythm-class distinction, English vs Dutch. The stimuli were natural forward utterances produced by multiple speakers. As predicted by the rhythm hypothesis, tamarins discriminated between Polish and Japanese, but not English and Dutch. These findings strengthen the claim that discriminating the rhythmic cues of language does not require mechanisms specialized for human speech. [Work supported by NSF.]

4pSC8. Infants long-term memory for complex music. Beatriz Ilari (Faculty of Music, McGill Univ., 555 Sherbrooke St. W., Montreal, QC H3A 1E3, Canada, bilari@po-box.mcgill.ca), Linda Polka (School of Commun. Sci. and Disord., McGill Univ.), and Eugenia Costa-Giomi (Faculty of Music, McGill Univ.)

In this study we examined infants' long-term memory for two complex pieces of music. A group of thirty 7.5 month-old infants was exposed daily to one short piano piece (i.e., either the Prelude or the Forlane by Maurice Ravel) for ten consecutive days. Following the 10-day exposure period there was a two-week retention period in which no exposure to the piece occurred. After the retention period, infants were tested on the Headturn Preference Procedure. At test, 8 different excerpts of the familiar piece were mixed with 8 different foil excerpts of the unfamiliar one. Infants showed a significant preference for the familiar piece of music. A control group of fifteen nonexposed infants was also tested and showed no preferences for either piece of music. These results suggest that infants in the exposure group retained the familiar music in their long-term memory. This was demonstrated by their ability to discriminate between the different excerpts of both the familiar and the unfamiliar pieces of music, and by their preference for the familiar piece. Confirming previous findings (Jusczyk and Hohne, 1993; Saffran *et al.*, 2000), in this study we suggest that infants can retain complex pieces of music in their long-term memory for two weeks.

4pSC9. Application of a model of the auditory primal sketch to cross-linguistic differences in speech rhythm: Implications for the acquisition and recognition of speech. Neil P. M. Todd (Dept. of Psych., Univ. of Manchester, Manchester M13 9PL, UK) and Christopher S. Lee (Univ. of Manchester, Manchester M13 9PL, UK)

It has long been noted that the world's languages vary considerably in their rhythmic organization. Different languages seem to privilege different phonological units as their basic rhythmic unit, and there is now a large body of evidence that such differences have important consequences for crucial aspects of language acquisition and processing. The most fundamental finding is that the rhythmic structure of a language strongly influences the process of spoken-word recognition. This finding, together with evidence that infants are sensitive from birth to rhythmic differences between languages, and exploit rhythmic cues to segmentation at an earlier developmental stage than other cues prompted the claim that rhythm is the key which allows infants to begin building a lexicon and then go on to acquire syntax. It is therefore of interest to determine how differences in rhythmic organization arise at the acoustic/auditory level. In this paper, it is shown how an auditory model of the primitive representation of sound provides just such an account of rhythmic differences. Its performance is evaluated on a data set of French and English sentences and compared with the results yielded by the phonetic accounts of Frank Ramus and his colleagues and Esther Grabe and her colleagues.

4:15–5:00

Panel Discussion

THURSDAY AFTERNOON, 6 JUNE 2002

BENEDUM ROOM, 2:00 TO 5:10 P.M.

Session 4pSP

Signal Processing in Acoustics and Underwater Acoustics: Advances in Sonar Imaging Techniques Including Synthetic Aperture Sonar and Computed Tomography

Brian G. Ferguson, Cochair

Maritime Systems Division, Defence Science and Technology Organization, P.O. Box 44, Pyrmont 2009, Australia

Kerry W. Commander, Cochair

Coastal Systems Station, Code R21, Panama City, Florida 32407-7001

Chair's Introduction—2:00

Invited Papers

2:05

4pSP1. Imaging sonar development for mine countermeasure applications. Bruce Johnson (Office of Naval Res., 800 N. Quincy St., Arlington, VA 22217)

Over the past decade the Office of Naval Research (ONR) has sponsored research to improve the mine countermeasures (MCM) community's ability to detect, classify, reacquire, and identify mines. In the first stage of wide area detection and classification, the single biggest problem is the ability to distinguish mines from clutter with side-scan sonar assets. In the reacquire-identify stage, divers employ a single-beam aural sonar for guidance to close proximity for visual identification of targets. Current sonar research efforts in wide-area search focus on clutter rejection through high-resolution imaging by synthetic aperture processing and cuing from broadband response characteristics. For reacquisition and identification, research has focused on compact, multibeam imagers. ONR's focus on transition from towbodies and divers to numbers of small, unmanned underwater vehicles (UUVs) to improve MCM clearance rates has had a major impact on sensor development as well. The presentation will provide an introduction to the session by providing a brief history of the development of navy imaging sonars, how they are employed, and an overview of the current thrusts that will lead to the next generation of Navy MCM sonars.

2:30

4pSP2. Application of computed tomography to the acoustic imaging of sea mines at safe standoff distances. Ron Wyber (Midspar Systems Pty. Ltd., 24 Farrer Pl., Oyster Bay NSW 2225, Australia) and Brian Ferguson (Defence Sci. and Technol. Organisation, P.O. Box 44, Pyrmont NSW 2009, Australia)

Typical imaging sonars make use of very high frequencies and arrays with a large number of sensors to provide high angular resolution. With such arrays the spatial resolution degrades and high propagation losses occur as the sonar is moved away from the target being imaged and operational ranges are generally 5 m or less. An alternative imaging technique has been implemented using computed tomography in which the target is insonified with a broad beam from multiple aspects. The spatial resolution achievable with this method is determined only by the bandwidth of the test signal and not by the beamwidth of the sonar or the distance of the sonar from the target. Results are presented showing clear images of mines and mine-like objects in which a spatial resolution of better than 2 cm is achieved at a range of 60 m, even for low target strength mines. Features are evident in the images that are related to both the internal and external shape of the targets as well as structural waves propagating within the targets. This offers the potential to provide unambiguous classification of mines from long ranges, even in a harsh operational environment.

2:55

4pSP3. Recent advances in synthetic aperture sonar for proud and buried object detection. M. Pinto, A. Bellettini, R. Hollett, and L. Wang (NATO SACLANT Undersea Res. Ctr., La Spezia, Italy)

This paper presents a comprehensive research program into synthetic aperture sonar (SAS) for proud and buried object detection. Experimental results are presented, which demonstrate the accuracy achievable with data-driven microneavigation (motion estimation) techniques, such as the displaced phase center antenna (DPCA) as well as aided inertial navigation systems (AINS). The combination of both methods is shown to provide a marked improvement in SAS performance compared with either method used alone. The impact of shallow and very shallow water environments is also discussed, using experiments specifically designed to study the effect of multipath and acoustic fluctuations. Finally, the issue of fast SAS image formation algorithms is addressed in the context of wideband/widebeam systems, nonlinear platform tracks, and sound velocity changes (as when sound penetrates into sediments). Preliminary results are presented for fast-factorized backprojection (FFBP), a promising algorithm derived for low-frequency SAR applications. [This work is done within the framework of a multinational Joint Research Program on Mine Detection and Classification and the authors would like to acknowledge the participation of all the Nations involved, in particular the UK and Norway.]

3:20

4pSP4. Synthetic aperture sonar imaging in a random ocean. Enson Chang (Dynam. Technol., Inc., 21311 Hawthorne Blvd., Ste. 300, Torrance, CA 90503-5602)

The maximum achievable aperture of a coherent imaging sonar is normally considered to be limited by the horizontal coherence length of the medium, which is on the order of several tens to several hundred wavelengths. Synthetic aperture sonar results from the past several years, however, have clearly exceeded this limit by an order of magnitude or more. It appears that the horizontal coherence length alone does not describe some important aspects of the phase errors present in real data. In particular, we suggest that underlying structures in the phase errors are exploitable in processing algorithms and are responsible for these results. We give a number of numerical examples from sub-kHz to tens of kHz, including new results on subcritical imaging of buried objects.

3:45–4:00 Break

4:00

4pSP5. Motion compensation technique for wide beam synthetic aperture sonar. Jose E. Fernandez, Daniel A. Cook, and James T. Christoff (Naval Surface Warfare Ctr., Coastal Systems Station, Dahlgren Div., 6703 W. Hwy. 98, Panama City, FL 32407)

Optimal performance of synthetic aperture sonar (SAS) systems requires accurate motion and medium compensation. Any uncorrected deviations from those assumed during the SAS beam formation process can degrade the beam pattern of the SA in various ways (broadening and distortion of the main lobe, increased side lobes and grating lobes levels, etc.). These would manifest in the imagery in the form of degraded resolution, blurring, target ghosts, etc. An accurate technique capable of estimating motion and medium fluctuations has been developed. The concept is to adaptively track a small patch on the sea bottom, which is in the order of a resolution cell, by steering the SAS beam as the platform moves in its trajectory. Any path length differences to that patch (other than the quadratic function product of the steering process) will be due to relative displacements caused by motion and/or medium fluctuations and can be detected by cross-correlation methods. This technique has advantages over other data driven motion compensation techniques because it operates in a much higher signal-to-noise beam space domain. The wide beam motion compensation technique was implemented in MATLAB and its performance evaluated via simulations. An overview of the technique, simulation, and results obtained are presented.

4p THU. PM

4:25

4pSP6. Acoustical imaging of high-frequency elastic responses of targets. Scot F. Morse (Div. of Computer Sci., Western Oregon Univ., Monmouth, OR 97361), Brian T. Hefner (Univ. of Washington, Seattle, WA 98105-6698), and Philip L. Marston (Washington State Univ., Pullman, WA 99164-2814)

Acoustical imaging was used to investigate high-frequency elastic responses to sound of two targets in water. The backscattering of broadband bipolar acoustic pulses by a truncated cylindrical shell was recorded over a wide range of tilt angles [S. F. Morse and P. L. Marston, "Backscattering of transients by tilted truncated cylindrical shells: time-frequency identification of ray contributions from measurements," *J. Acoust. Soc. Am.* (in press)]. This data set was used to form synthetic aperture images of the target based on the data within different angular apertures. Over a range of viewing angles, the visibility of the cylinder's closest rear corner was significantly enhanced by the meridional flexural wave contribution to the backscattering. In another experiment, the time evolution of acoustic holographic images was used to explore the response of tilted elastic circular disks to tone bursts having frequencies of 250 and 300 kHz. For different tilt angles, specific responses that enhance the backscattering were identified from the time evolution of the images [B. T. Hefner and P. L. Marston, *Acoust. Res. Lett. Online* **2**, 55–60 (2001)]. [Work supported by ONR.]

4:40

4pSP7. Improved ping-to-ping cross-correlation methods for synthetic aperture self-calibration. Didier Billon and Franck Fohanno (Thales Underwater Systems, B.P. 60114, 29601 Brest Cedex, France)

Ping-to-ping cross-correlation (P^2C^2) methods, like DPC (Displaced Phase Center), are an efficient way to estimate motion-related parameters for synthetic aperture beamforming, as shown by several sonar sea trials performed recently. Their accuracy depends much on the space sampling rate of the synthetic aperture. Achieving large resolution gain implies decreasing by a factor up to 2 the travel speed with respect to the absolute upper limit $L/2T$. The paper reports on two new P^2C^2 schemes that

greatly relax this constraint. "Auxiliary frequencies P^2C^2 " performs on signals from two additional frequencies transmitted at two electronically swapped phase centers close to both real aperture ends, simultaneously with the main frequency used to form the synthetic aperture. "Hybrid P^2C^2 " consists in estimating bearing shift by means of a 2D gyrometer and an auxiliary vertical array, rather than from the main horizontal array signals. The principles of these processings are described and analytical expressions for performance bounds are given. Sonar images from a new towed sidescan sonar for deep sea floor mapping, obtained by means of both new P^2C^2 schemes, are presented. 25 cm cross-range resolution is achieved up to 800 m range. Then the resolution gain exceeds 30.

4:55

4pSP8. Extending the capabilities of synthetic aperture sonars (multiaspect and multiband) in the overall objective of buried mines classification. Alain P. Hetet, Benoit Zerr (Groupe D'Etudes Sous-Marines L'Atlantique, BP 42-29240 Brest Naval, France), and Pascal Challande (Laboratoire de Mecanique Physique, Universite Paris VI, St. Cyr, France)

Low- and medium-frequency sonars are more adequate for long-range detection and classification operations needed for mine hunting. In addition, with low frequencies the acoustic waves can penetrate into the sediment. With such a technique, detection of buried mines is possible. But, in real conditions, at sea, the number of alarms and false alarms is important. One of the problems is to find solutions to reduce those false alarms. One technique developed in this presentation is the multiband technique in association with large-bandwidth acoustical arrays. The aim of this technique is to reduce the speckle effect with noncoherent summation of multiband coherent SAS images. This technique has been focused, at the moment, on mines laying on the seabed, and is expected to be used on buried mines. The second technique discussed in the presentation is the multi-aspect SAS processing applied in the low-frequency case for buried mines detection and classification. This technique is intended to help during the echo classification process of buried mines, limiting the false-alarm rate and reconstructing the shape of the object by following the echo along the shape of the objects.

THURSDAY AFTERNOON, 6 JUNE 2002

KINGS GARDEN SOUTH, 2:00 TO 4:20 P.M.

Session 4pUW

Underwater Acoustics: Shallow Water Environmental Acoustics: Session in Honor of Morris Schulkin

Robert C. Spindel, Chair

Applied Physics Laboratory, University of Washington, 1013 Northeast 40th Street, Seattle, Washington 98105-6698

Chair's Introduction—2:00

Contributed Papers

2:05

4pUW1. Comparison of ASIAEX South China Sea acoustic fluctuations with oceanographic variability. Timothy F. Duda, James F. Lynch, and Arthur E. Newhall (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

During May 2001, signals were transmitted from five moored 16- to 100-Hz-bandwidth sources to collocated horizontal and vertical receiving arrays in 120-m-depth water on the continental slope. The site was in the northern portion of the South China Sea. Transmissions were nominally 20 km cross slope to the receiver from a site with three closely spaced

sources, and 32 km up-slope to the receiver from a site with two sources, although the bathymetry was complex. Signal frequencies ranged from 210 to 550 Hz. A broad spectrum of water column variations was observed during the 17-day acoustic experiment with our array of moorings at 20 sites, including the source and receiver locations. Internal tides were consistently present, as were nonlinear internal waves of moderate amplitude. Nonlinear internal waves of larger amplitude propagating from the eastern South China Sea were present only episodically. Subtidal frequency water-mass variations also occur. Correlation between various measures of the acoustic signal variability and the water column variability will be examined. [Work supported by ONR.]

2:20

4pUW2. Depth averaged acoustic field variability—its relationship to sound speed field variability—comparison of SWARM95 data to PE calculations. Peter Mignerey and Marshall Orr (Acoust. Div., The Naval Res. Lab., 4555 Overlook Ave. S.W., Washington, DC 20375)

Depth and time averaged acoustic signal energy was found to change by about 3 db over a 2.3 day period during an experiment on the New Jersey Shelf (SWARM95). The sound speed field along the 42 km propagation path has been reconstructed by integrating point CTD and range continuous acoustic flow visualization data taken during four transects of the acoustic propagation path. Sound speed fields were constructed for each transect. Depth averaged acoustic signal energy calculated for each of the sound speed fields are compared to the measured depth averaged acoustic signal energies.

2:35

4pUW3. Estimates of the prevalence of anomalous signal losses in the Yellow Sea derived from acoustic and oceanographic computer model simulations. Stanley A. Chin-Bing, David B. King, Alex C. Warn-Varnas (Naval Res. Lab., Stennis Space Center, MS 39529-5004, chinbing@nrlssc.navy.mil), Kevin G. Lamb (Univ. of Waterloo, Waterloo, ON N2L 3G1, Canada), James A. Hawkins (Planning Systems, Inc., Slidell, LA 70458), and Marvi Teixeira (Polytechnic Univ. of Puerto Rico, Hato Rey 00988, PR)

The results from collocated oceanographic and acoustic simulations in a region of the Yellow Sea near the Shandong peninsula have been presented [Chin-Bing *et al.*, *J. Acoust. Soc. Am.* **108**, 2577 (2000)]. In that work, the tidal flow near the peninsula was used to initialize a 2.5-dimensional ocean model [K. G. Lamb, *J. Geophys. Res.* **99**, 843–864 (1994)] that subsequently generated internal solitary waves (solitons). The validity of these soliton simulations was established by matching satellite imagery taken over the region. Acoustic propagation simulations through this soliton field produced results similar to the anomalous signal loss measured by Zhou, Zhang, and Rogers [*J. Acoust. Soc. Am.* **90**, 2042–2054 (1991)]. Analysis of the acoustic interactions with the solitons also confirmed the hypothesis that the loss mechanism involved acoustic mode coupling. Recently we have attempted to estimate the prevalence of these anomalous signal losses in this region. These estimates were made from simulating acoustic effects over an 80 hour space-time evolution of soliton packets. Examples will be presented that suggest the conditions necessary for anomalous signal loss may be more prevalent than previously thought. [Work supported by ONR/NRL and by a High Performance Computing DoD grant.]

2:50

4pUW4. Influence of environmental parameters on mid-to-high frequency acoustic signal propagation in coastal region^{a)} Luc Lenain, Mohsen Badiey (Ocean Acoust. Lab., College of Marine Studies, Univ. of Delaware, Newark, DE 19716), and Steve E. Forsythe (Naval Undersea Warfare Ctr., Newport, RI 02341)

Broadband mid-to-high frequency (0.6–18 kHz) acoustic wave propagation in shallow coastal waters (20 m) is influenced by a variety of oceanographic conditions. Physical parameters such as temperature and salinity as well as hydrodynamic parameters such as surface waves, tide, and current can influence amplitude and travel time of acoustic transmissions. While some of these physical parameters can be extracted from the acoustic wave propagation, understanding the forward model is vital to a fundamental understanding of the inverted parameters in shallow coastal waters. A unique set of simultaneous ocean and acoustic observations that reveal interesting temporal behavior of the acoustic signal and its correlation with environmental variability are presented. Some forward modeling using full-wave and ray-based methods is employed to investigate the time-frequency-angle characteristics of the measured acoustic field. Re-

sults of acoustic inversion for current, salinity, and sea-surface roughness, as well as the limitations of such inversions when using broadband signals in very shallow water regions, are also discussed. [Work supported by ONR and Sea Grant.] ^{a)}For Underwater Acoustics Best Student Paper Award.

3:05–3:20 Break

3:20

4pUW5. Sound scattering in shallow water in presence of internal solitons. Boris G. Katsnelson, Serguey A. Pereselkov (Voronezh Univ., 1 Universitetskaya Sq., Voronezh 394693, Russia), and Valery G. Petnikov (Gen. Phys. Inst., 117333 Moscow, Russia)

Sound scattering by localized inhomogeneity (object) in a shallow-water waveguide is studied. Influence of internal solitons (IS) traveling in the sea, on the scattered field is considered. The problem is analyzed within the framework of theory of the sound scattering in the waveguide developed by the authors, where the scattering matrix is expressed through modal decomposition including the scattering amplitude of the object in free space. The peculiarity of the given work is the taking into account of additional modes conversion due to IS. The waveguide and IS are modeled on the basis of experimental data (sound-speed profile, bottom relief, IS parameters, etc.) collected in the Japan Sea. The inhomogeneity is selected in the form of soft spheroids with dimensions characteristic of gray whales (*esrichtius*). Calculations of amplitude fluctuations of the low-frequency sound field at the receiving point are carried out for moving spheroid and different orientations of acoustic track with respect to direction of propagation of IS. It is shown that sound fluctuations have essentially different characteristics for longitudinal and transversal propagation of IS. [Work supported by RFBR, Grant No. 00-05-64752.]

3:35

4pUW6. Impacts of internal tides on acoustic propagation at sonar frequencies and their effects on sonar system performance. Subramaniam D. Rajan, James K. Lewis, Jason Rudzinsky, and Peter J. Stein (Sci. Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, srajan@scisol.com)

Temperature oscillations triggered by ocean tides can be found along nearly all shelf-slope regions throughout the world's oceans. It is of interest to ascertain how these internal tides impact acoustic propagation for operational purposes. Internal tides have been observed around the Hawaiian Islands, and these tides result in large fluctuations in temperature on a semidiurnal basis. Ocean models are presented that reproduce tidally induced sound speed variations (through model-predicted variations in temperature and salinity) around the islands of Oahu and Kauai, Hawaii. The model provides sound-speed structure and variability, and this is used as input to acoustic propagation models. Model-predicted sound-speed structures are functions of space and time. We present results of studies carried out to investigate the effect of the internal tides on the performance of sonar systems. We quantify these effects by studying the impact of internal tides on factors such as the detection range to target in active systems and source localization errors in passive systems. A means of quantifying uncertainty in the performance of the sonar systems due to the presence of the internal tides is investigated by studying the performance of the sonar system by varying the strength of the internal tidal field.

4pUW7. Challenges and possible solutions to acoustic tomography in shallow-water areas. Jason Rudzinsky, Peter J. Stein, Subramaniam D. Rajan, and James K. Lewis (Scientific Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, jrudzinsky@scisol.com)

The Pacific Missile Range Facility off the island of Kauai in the Hawaiian chain is equipped with a large number of bottom-mounted sources and receivers. This instrumentation provides an immense opportunity for the ocean acoustic community to create an ocean acoustic laboratory. Such a laboratory can be used to monitor the changes in the three-dimensional sound-speed structure in the region using acoustic tomography. An ocean model developed for the region predicts the sound-speed structure of the region given the atmospheric, tidal, and mesoscale circulation forcing functions. This information, supplemented by acoustic tomography results, can be used to develop a consistent picture of the sound-speed field of the region. We present the challenges involved in successful implementation of tomography. The challenges are posed by the location of the sources and receivers, the bathymetry of the region, the nature of the ocean/bottom interface, the limitation on the bandwidth of the source signal, and the factors affecting the acquisition of data. We will also present approaches

that can be taken to overcome some of the problems. Preliminary results of a field experiment conducted to assess the feasibility of tomography in the region will be presented.

4pUW8. Propagation of sound in the surf zone. Gary Sammelmann (Code R22 Coastal Systems Station, 6703 W. Hwy. 98, Panama City, FL 32407-7001), Grant Deane (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA 92093-0238), Svein Vagle, and David Farmer (Inst. of Ocean Sci., Sidney, BC V8L 4B2, Canada)

This presentation focuses on the effects of a time-varying sea surface and bubble clouds due to breaking waves on the propagation of sound in the surf zone. The rough sea surface generates caustics corresponding to Bragg diffraction due to the focusing of sound by the rough surface as described by the specular point method. A comparison with experimental data taken by Grant Deane is presented. Time evolution of bubble clouds in the surf zone is modeled by the algorithms developed by Svein Vagle, David Farmer, and Grant Deane. Bubbles in the surf zone alter both the attenuation of sound and the sound-velocity profile in the ocean. [Work supported by Codes 321OE and 32MIW of the ONR.]